THE UNIVERSITY OF ALBERTA

SPEECH COMPRESSION BY SMALL COMPUTER

by

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FACULTY OF GRADUATE STUDIES

The undersigned certify that they have read, and recommend to the Faculty of Graduate Studies for acceptance, a thesis entitled "Speech Compression by Small Computer" submitted by Shahid Ul Haq Qureshi in partial fulfilment of the requirements for the degree of Master of Science.

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FACULTY OF GRADUATE STUDIES

The undersigned certify that they have read, and recommend to the Faculty of Graduate Studies for acceptance, a thesis entitled "Speech Compression by Small Computer" submitted by [illegible] in partial fulfilment of the requirements for the degree of Master of Science.

ABSTRACT

This report describes the implementation, on a PDP-8 computer, of a scheme to compress speech by removing its temporal redundancy. Three features of the speech waveform are extracted for every 12.8 msec segment while speech is being digitized and stored. A decision making program determines the contiguous segments which can be grouped together on the basis of these features. Measurements on the waveform are also used to identify the type of speech sound represented by each group of segments. Based on this information redundant segments are found and the output program is instructed to skip these during digital to analog conversion. The thresholds for decision making are continually adjusted according to the intensity level of the utterances. Care is taken to minimize the transients in the vicinity of the junctions of the retained segments. With 4K words of core capacity and a disk memory of 32 K words, short sentences have been compressed. The method of compression developed is selective and minimizes processing time on the computer. The resulting compressed speech is fairly intelligible and of good quality.

ACKNOWLEDGEMENTS

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INTRODUCTION

The importance of spoken language as a means of communication in daily life can hardly be over emphasized. As in visual reading, a significant variable in aural communication is the rate at which it occurs. This is of special interest to those who, for one reason or another, must depend upon aural communication.

Time compressed speech is speech which has been reproduced in less than the original production time, that is, speech at an increased rate. Apart from being useful in various educational settings, compressed speech may be employed in studying the temporal requirements of the listener as he processes spoken language (1).

The most obvious method of increasing word rate is speaking rapidly. This method has serious drawbacks in that the speaker must be well trained and even then the rate and the clarity are limited by the physical processes involved in speech production.

Other methods of speech compression take advantage of the fact, indicated by early speech research, that much of the natural speech signal is redundant (2). It was found that the speech wave pattern could be variously distorted without serious losses in the resulting intelligibility. Fletcher (3) studied the intelligibility of speech compressed by reproducing a tape or record at a speed faster

than the one used during recording. Losses in intelligibility were found to be small until the speed was 1.4 times that of the original speech speed. This method is limited, however, by the accompanying distortion due to the frequency shift.

In 1950, Miller and Licklider (4) showed that speech remained intelligible if interrupted more than ten times a second until about half of the original speech signal had been removed. Based on this work Garvey (5) obtained compressed speech by removing the silent spaces in the interrupted speech and splicing together the remaining segments of the tape record. This "chop-splice" technique resulted in compressed speech with no frequency distortion and with reasonable intelligibility. One year later Fairbanks, et al, (6) described an electro-mechanical apparatus for time compression or expansion of speech which used the general principle demonstrated by Garvey. Similar approaches had previously been indicated by Gabor (7,8) and others. This method of speech compression has come to be known as the sampling method since recorded speech is sampled by retaining and discarding portions of the speech periodically. It is obvious that the sampling method is unselective with respect to the portions of a recorded signal that are discarded. There is, therefore, some probability that the discarded segments may contain auditory cues essential to perception. The probability that an auditory cue may lie entirely within a discarded segment decreases as the discard interval is made smaller. However, when the discard interval is small the recorded speech has to be sampled more often to obtain the same amount of compression. If the sampling frequency becomes high enough to be audible it interferes badly

with the speech signal.

Another device for speech compression, the "Harmonic Compressor", based on research carried out at the Bell Telephone Laboratories, Inc. (9), has been developed at the American Foundation for the Blind. In this device the speech signal is separated into individual harmonic frequency components by an elaborate bank of bandpass filters and the frequencies are halved. This half-spectrum speech is then resynthesized and recorded. Playing the record at twice the speed used during recording restores the frequency spectrum, resulting in speech compressed to 50% in time without any shift in pitch. A serious limitation of the harmonic compressor is that it cannot be adjusted for any desired amount of compression. Moreover, unvoiced sounds and noise, which are devoid of the quasi-periodicity on which the harmonic compressor is based, are distorted to some extent.

Compressed speech can also be produced by actually synthesizing speech at a rate faster than normal (10). This is accomplished by recording the control parameters of speech obtained from a vocoder analyzer and supplying them at a faster rate to a vocoder synthesizer to remake compressed speech. It would also be possible to synthesize faster speech by rule on a digital computer. This method has, as yet, received little development mainly because it is the most expensive to implement.

Digital computers have been used to compress speech in a number of ways. Fairbanks' sampling method can be easily simulated on the computer (11), and the durations of the retained and discarded

segments can be varied over a wide range. In 1966 Scott (12,11) proposed the dichotic method of speech compression on a computer. In this method, speech compressed by the sampling method is presented to one ear and the discard intervals are joined sequentially and presented to the other ear. Dichotic speech appears to have some advantage over speech compressed by the sampling method when reproduced in more than 50 per cent of the original production time (13). The superiority, however, is too small to be of practical significance (14).

Another method of compression attempted on the digital computer is the pitch period compression of speech (11,12,15). The locations of the pitch periods are calculated and a number of pitch periods are discarded depending upon the desired compression. Unvoiced sounds can be left alone or a discard interval can be arbitrarily established such as the average of the detectable periods in the immediate vicinity. The quality of speech compressed by this method depends on the level of sophistication employed in the pitch period detection process (which is known to be a difficult task (2)). The cost is enormous, perhaps 300 dollars per minute of original speech (15).

From this brief review of the methods of speech compression it is obvious that the duration of certain phonemes in normal speech is longer than required for reliable recognition. In other words parts of some of the speech sounds are redundant. The object of speech compression is to remove this temporal redundancy and thus convey more information in less time. Although unselective removal of portions of recorded speech results in compressed speech a satisfactory method must be selective. Such a method would remove only those parts of the speech

signal which are redundant from the point of view of perception of the speech sounds. Although the high cost of computer time on large computers is a prohibitive factor in the use of computers for speech compression, the demands of a selective method are best met by a digital computer. As described earlier the digital computer has been used in a number of ways for compressing speech but a selective or differential method has not yet been reported in literature (16).

The object of this project has been to find an economical way of compressing speech on a computer, and to develop a selective method of compression utilizing the great flexibility offered by a programmed data processor. The possibility of compressing speech on a small computer, the PDP-8, has been investigated with a view to developing a satisfactory method with a minimum of processing time on the computer.

The first step in selective speech compression is to distinguish between various speech sounds. A number of methods of speech segmentation have been developed for the automatic recognition of speech (17,18,19). The purpose of the segmentation process in this case, however, is to locate redundant parts rather than find sharp boundaries between phonemes. A time-domain method similar to (19) is used in preference to others (involving comparison of spectral properties) to avoid costly hardware or excessive computer time in finding the spectrum.

The segments of the speech signal are first classified into transitional and sustained segments. Sustained segments are those which possess certain features of the speech waveform which do not change appreciably over the duration of the segment. These segments are also tagged as vowel-like, fricative, plosive or silence according to the

properties of the speech waveform. Final decisions are then made as to what parts of the speech signal can be removed without adversely affecting the perceptual cues contained in the signal. The decisions also take into consideration the desired amount of compression. The remaining speech segments are then abutted in time while carefully minimizing the transients at the junctions of the segments.

The segmentation and classification processes used are an attempt at achieving a compromise between sophistication and large processing time on the computer. The level of sophistication achieved appears to be sufficient for the purpose of the problem at hand.

On the present set-up with 4 K words of core capacity and a disk memory of 32 K words, speech has been processed only a sentence at a time. The limitation is due to the small storage capacity of the disk.

All the programming of the PDP-8 computer was done in machine language using the mnemonic operation codes. PAL-D Assembler (Program Assembly Language for the Disk system) was used to assemble and translate the source program statements into the binary codes needed in machine instructions. Machine language programming, though cumbersome, was chosen to make an efficient use of processing time and the storage space available in the computer.

The whole process of speech compression can be divided into three steps: 1) Input and Feature Extraction, 2) Decision Making and 3) Transient Removal and Output. The following three chapters describe each of these steps in some detail.

CHAPTER II

INPUT AND FEATURE EXTRACTION

Time compression cannot be done in real time, because this would amount to predicting what the speaker was about to say. Therefore the speech signal to be compressed must be available to the device in recorded form regardless of what device is used for compression. In the case of a digital computer as a compressor, speech must be fed in and stored in a form suitable for use by the computer.

Fig. 1 is a block diagram representation of the system used for this project. The digital computer used is a standard PDP-8 with a core capacity of 4 K words of 12 bits each. The teletype and DEC tape units connected to the computer were used for software development only. The DF 32 disk memory of 32 K words provided the storage for the digitized speech and the programs. The interface consisted of REDCOR 12-bit analog to digital and digital to analog converters and logic circuits for timing the operations from an external clock (in this case a GR pulse generator).

Input

Recorded speech, reproduced at one-half the speed used during recording, is band-limited to 2.5 KHz and digitized at 5 KHz to obtain an effective sampling rate of 10 KHz. The sampling and digitization is carried out by the 12-bit A/D converter. The ordinate of the slowed down speech waveform is read every 200 μ secs and quantized to the near-

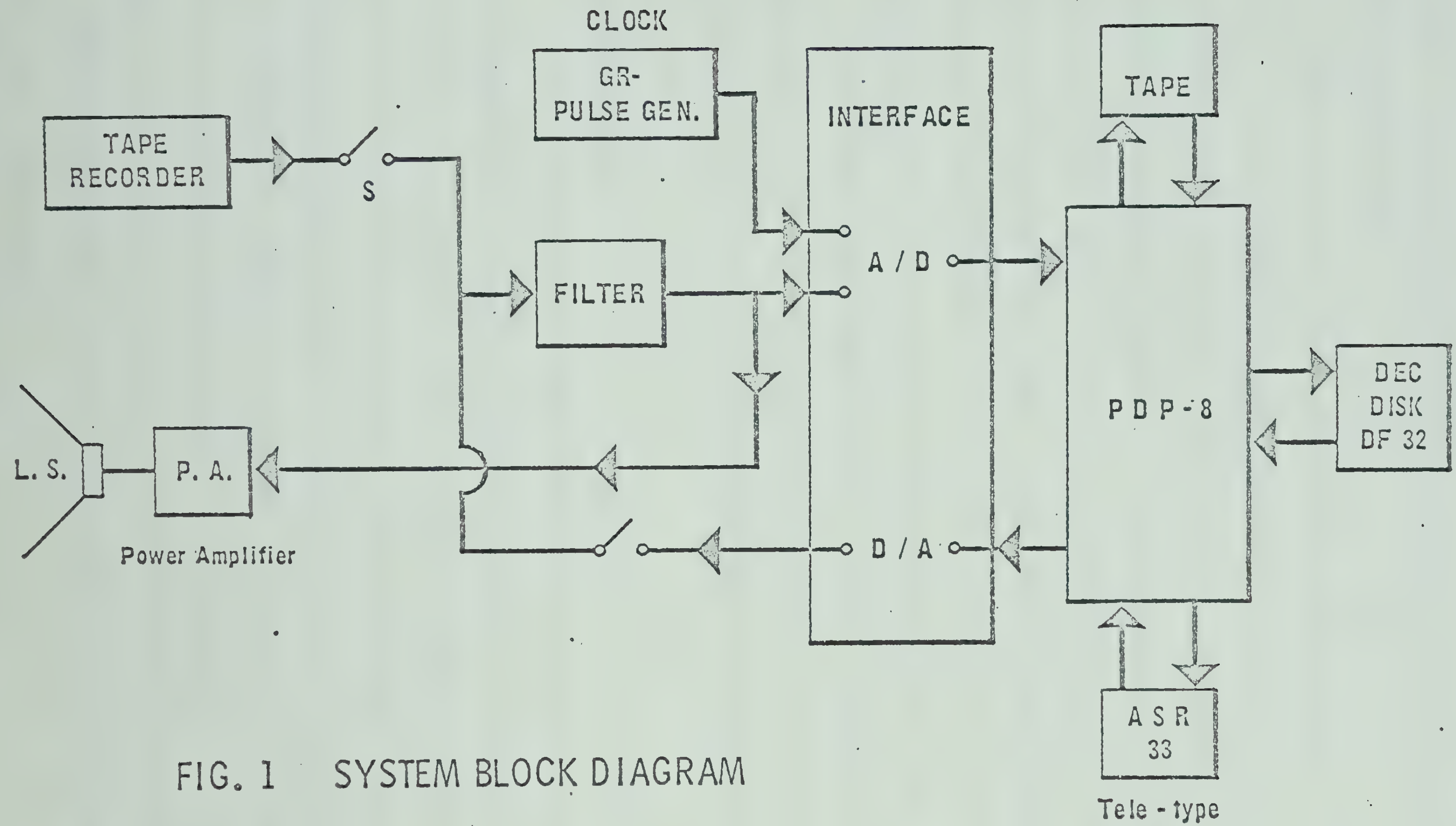

FIG. 1 SYSTEM BLOCK DIAGRAM

est one of the 4096 (2^{12}, 2048 positive and 2048 negative) levels. The samples are temporarily stored in core and then swapped on to the disk in large blocks of data.

The speech input takes place under program control. Three features of the speech waveform are extracted while speech is being digitized and stored. The start of the program is triggered by the speech signal from the tape recorder. This synchronization of the start of the speech signal and the program allows full use of the storage capacity of the disk to be made. Any silence interval before the speech signal is not digitized or stored. An electronic comparator and a skip logic circuit in the interface were used to achieve this control.

The speech input continues until the disk memory is full (approximately three seconds of speech). Control is then transferred to a subroutine which reads the decision-making and output programs from the disk.

Feature Extraction

As a first step in feature extraction the speech wave is divided into segments the duration of which corresponds to 12.8 msecs of speech played back at normal speed. The duration of the segment was chosen to be large enough to include at least one complete pitch period of the heavy male voice and small enough so that significant changes do not occur during the segment. The particular value of 12.8 msecs was selected for ease of handling segments of 128 samples on the PDP-8 computer, the core memory of which is organized into pages of 128 words each. The following three features are extracted for every segment and stored in the computer core:

1) The sound intensity 'I' defined as the absolute maximum of 128 samples, the samples being the ordinates of the speech waveform at constant intervals of 100 μ secs. If the 128 samples are represented by a vector Y then sound intensity

$$I = \max |y_i|, \quad i = 1,2,\ldots,128$$

y_i being the elements of Y.

2) The waveform asymmetry (20) 'A' defined as the difference between the positive maximum and the negative maximum of 128 samples.

$$A = \max y_p - \max y_n$$

where y_p includes all the positive elements of Y

y_n includes all the negative elements of Y

3) The number of zero crossings 'Z' in 128 samples. A zero crossing is said to occur whenever the sign of the ith element of Y is different from the sign of the $(i + 1)$th element, i.e., if

$$y_i \cdot y_{i+1} = - |y_i| |y_{i+1}|$$

The above characteristics of the speech waveform were chosen for the simplicity with which they can be extracted from raw speech. They are also quite effective in segmenting and identifying different types of sounds. An attempt was first made to compute the power spectra of speech at 10 msec intervals using the fast Fourier transform algorithm. It was planned to use spectral properties, such as ratio of power in different bands, for segmentation. This approach was soon dropped in favour of the one described here. The reason was the excessive computer time (of the order of 1 sec for 50 msecs of speech) taken in computing the spectrum on a small machine like the PDP-8.

The time-domain approach of segmentation has been used by others. Sakai and Doshita (21) used zero-crossing wave analysis and information about voicing to separate vowel-like phonemes from others. Different vowel-like phonemes were further distinguished by formant-stability criteria obtained from zero-crossing analysis. Hughes and Hemdal (22) used information about voicing, silence and turbulence together with the property that semivowels and nasals are less intense sounds compared with the neighbouring vowels. In the scheme used by Reddy (19) the segmentation is mainly based on the variation or stability of sound intensity levels. He uses zero-crossing counts for error correction.

A glance at the speech waveform, Fig. 2, will show that the sound intensity, or the amplitude of the envelope, varies considerably during vowel-consonant and consonant-vowel transitions. The intensity level does not however vary much during the quasi-stable vowel-like sounds. This feature of the speech waveform is thus a good measure of the stability of vowel-like sounds. The sound intensity can also be used for separating semivowels and nasals from vowels because of the difference in the intensities of these sounds. Fricatives and plosive bursts are the sounds with the lowest amplitude and can thus be distinguished from vowel-like sounds. The sound intensity is also an indicator of the presence or absence of voicing and is effective in separating pause intervals from phonation.

It is known, and it was verified experimentally, that infinitely clipped speech, a rectangular zero-crossing wave, is fairly intelligible. This leads us to believe that much of the information of the original speech signal is preserved even after amplitude simplification.

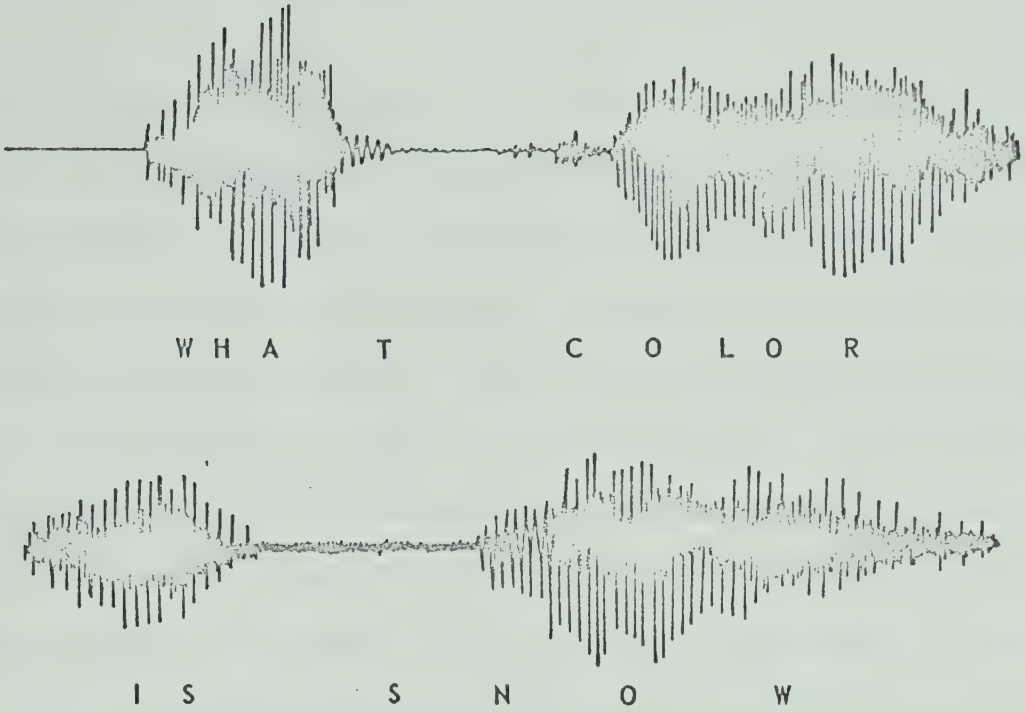

FIG. 2 SOUND INTENSITY VARIATIONS IN SPEECH WAVEFORM

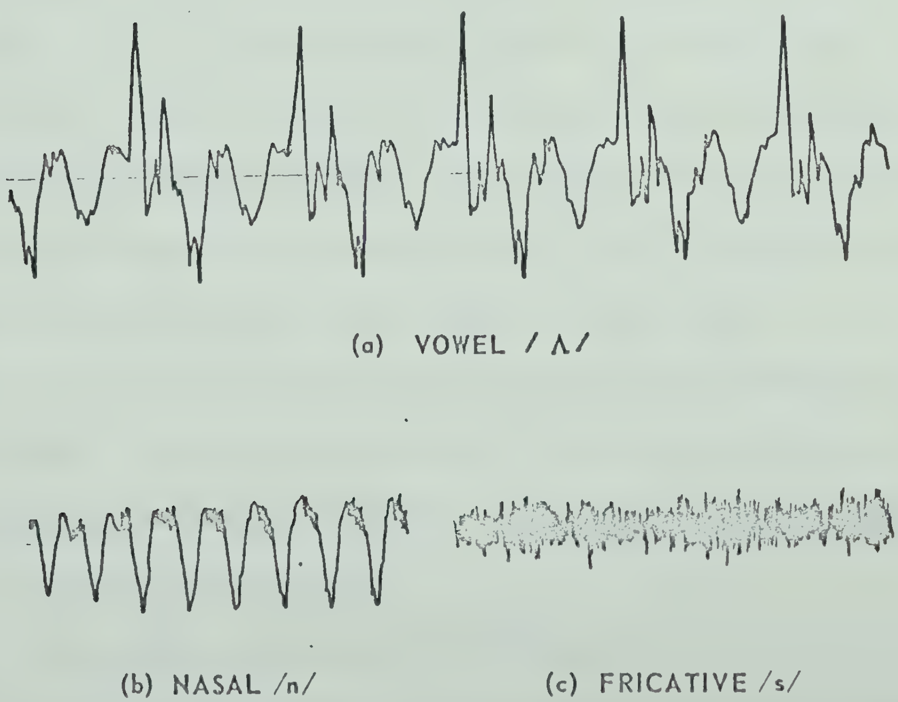

FIG. 3 WAVEFORM ASYMMETRY IN VOICED AND UNVOICED SOUNDS

This information is contained in the width of each rectangular wave between zero-crossings. It was used effectively by Sakai and Doshita (21) and others. A simplified characteristic, the zero-crossing count, is used here for ease of extraction. Though not as effective as the zero-crossing width analysis, when used alone, it does give an indication of the frequency content of the speech sound. When used with the other two features it helps in distinguishing fricatives and plosive bursts from other types of sounds and silence intervals, apart from being an additional measure of the stability of any type of sound.

Waveform asymmetry was first used for identifying and classifying voiced sounds in a 15-word vocabulary, voice-controlled adding machine developed by IBM Corporation, called Shoebox (23). The results of more recent work on the effectiveness of asymmetry measurement in identifying voiced sounds have been reported by Comer (20). All voiced sounds exhibit asymmetry, that is, there is a difference between the positive and negative peaks of the speech waveform. Unvoiced sounds, however, are composed of nonharmonically related components and are symmetrical about the base line. This is illustrated in Fig. 3. The value and polarity of the waveform asymmetry also varies for different voiced sounds. This characteristic of the speech waveform can thus be employed to identify voiced sounds as opposed to unvoiced sounds and to segment different voiced sounds.

In addition to these three features the location of the positive peak in each 12.8 msec segment is found and stored in the computer core during input. The locations of the positive peaks are later used to help remove transients that occur due to the deletion of redundant segments.

The Program

The programs for each step in the speech compression process are designed to work independently. They are initially stored on the disk and are called sequentially into the main core memory of the computer and executed. The input and feature extraction program consists of instructions for A/D conversion, for writing the speech samples on the disk memory and for extracting the three features and the locations of the positive maxima. The flow-chart and core usage diagram for this program are shown in Fig. 4 and Fig. 5 respectively. The program starts reading from the A/D channel as soon as it senses a pulse indicating the start of the speech signal. The program loop is timed by pulses from the pulse generator so that A/D conversion takes place at constant intervals. The digitized speech samples are stored in a large area in core and later swapped onto the disk periodically without interfering with the A/D conversion. This is possible due to the data-break facility, for data transfer to the disk, available in the system. The instructions for feature extraction are also present in the same loop and the three features plus the location of the positive maximum are stored after each segment of 128 words has been read in.

After modification in only one location the program is capable of sampling at 10 KHz so that slowing the speech signal would not be necessary. This will, however, introduce slight errors in the sampling interval after each segment and whenever the subroutine for writing on the disk is executed. The length of the program loop at these points causes the sampling interval to be larger than 100 μ secs for one sampling period. The effect of the error, however, is negligible.

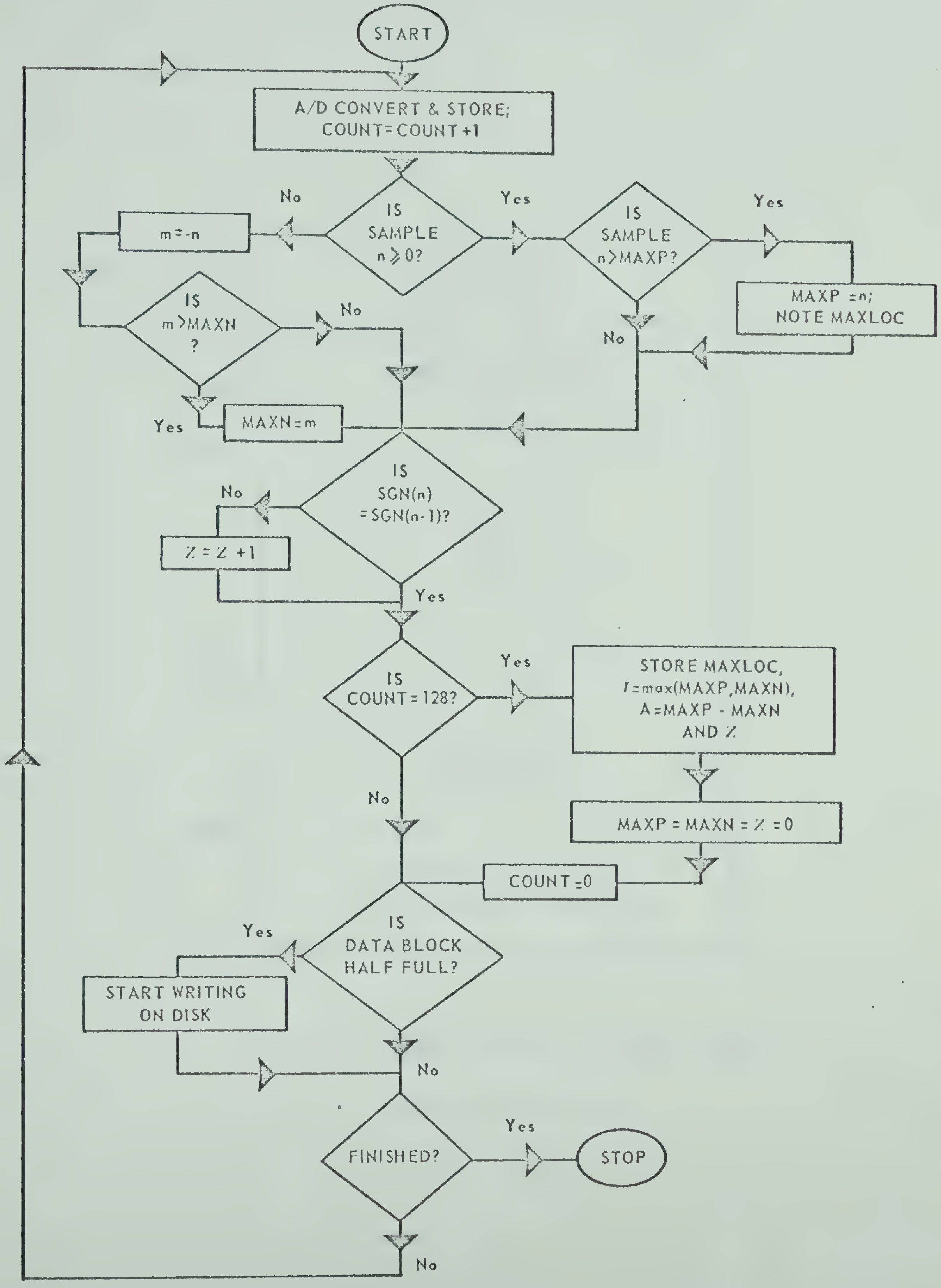

FIG. 4 FLOW-CHART FOR INPUT AND FEATURE EXTRACTION PROGRAM

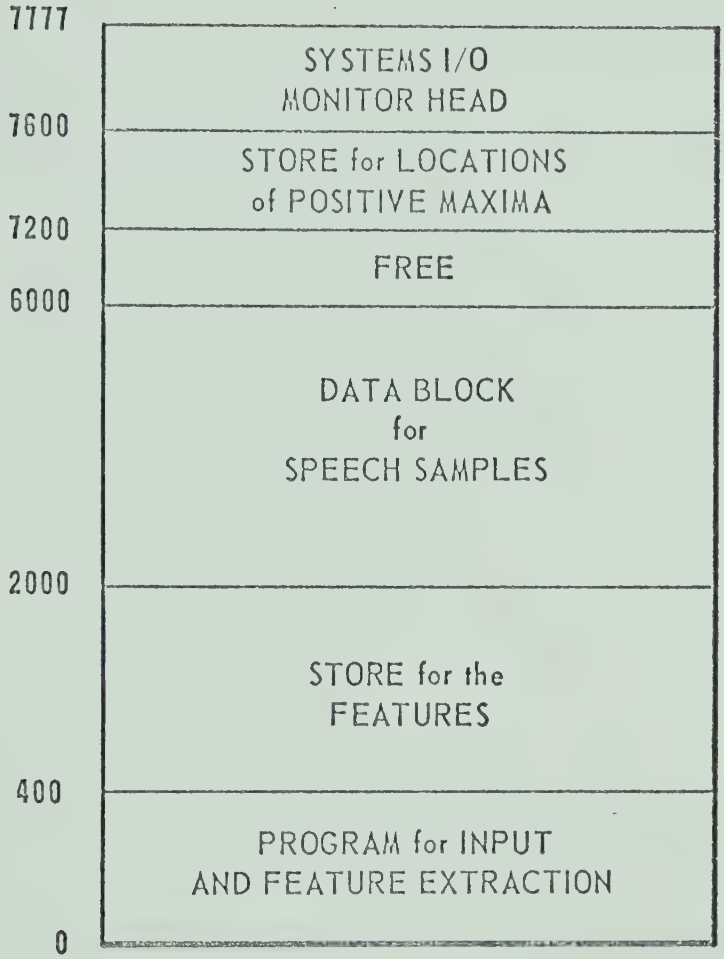

FIG. 5 CORE USAGE DIAGRAM FOR INPUT PROGRAM

At the end of the input the next program is automatically read into a vacant area in core. Control is then passed on to this program for the most crucial step in the compression process: decision making.

CHAPTER III

DECISION MAKING

Once speech has been digitized and stored and useful information about the speech waveform has been extracted, decision making is the next logical step. In this step decisions, based on the available information, are made as to which portions of the speech signal can be discarded without adversely affecting its perception. The decision making program accepts as input the three features of the speech waveform discussed in Chapter II and produces an output indicating which particular 12.8 msec segments are to be discarded.

In the first part of this program each of the three features for one segment is compared with those of the contiguous segments. An attempt is made to find similar segments which can be grouped together to represent a sustained speech sound. This is based on the reason that in the time domain the speech waveform can be thought to be composed of two types of intervals: a) quasi-stable intervals in which the parameters remain in an almost constant state; b) transitional intervals in which the parameters change gradually except for some time points at which parameters change abruptly. Thus the initial segmentation procedure involves the location of time intervals during which no significant change takes place.

The values of two of the three features, namely, the "sound intensity" and the "waveform asymmetry", depend on the amplitude of the

utterance. Fixed thresholds for the comparison of these features would, therefore, result in amplitude dependent decisions. The three more obvious methods of overcoming this problem are: a) to pass the speech signal through an appropriate automatic gain control device before it is A/D converted; b) to normalize the amplitude of the digitized samples in the digital computer and c) to continually adjust the thresholds according to the amplitude of the speech signal. The first of these needs extra hard-ware and the second takes excessive computer time. The third, though not as sophisticated as the first two, was chosen because it is the most economical to implement. The thresholds for decision making (Table 1) are set every 128 msecs, their values depending upon the intensity level of the utterance during the next 410 msecs. The decisions are thus made amplitude insensitive.

A flow-chart for the program is shown in Fig. 6. The thresholds are first set and a test is carried out to determine whether the current and the next segment can be grouped as silence. Two segments are grouped as silence or pause if the sound intensity of the nth segment $I_n \leq$ SILENCE (Table 1) and $I_{n+1} \leq$ SILENCE and if the number of zero crossings in the nth segment

$$Z_n \leq 16 \text{ and } Z_{n+1} \leq 16$$

If the test for silence succeeds the group is tagged as "silence" and the program proceeds to test the next segment. In this case the subroutines for comparing the three features are bypassed. Care is taken to start a new group of segments whenever there is a transition from a segment that is "silence" to one that is "not silence" and vice versa.

If the test for silence fails the subroutines for comparing "waveform asymmetry", "sound intensity" and "number of zero crossings"

TABLE 1

THRESHOLDS FOR DECISION MAKING

I_{max} = maximum sound intensity in 410 msecs of speech

Threshold	Value
1. "Voiced-Unvoiced" threshold	12 db below I_{max}
2. "W" threshold for waveform asymmetry	18 db below I_{max}
3. Threshold for "SILENCE"	22 db below I_{max}
4. Minimum tolerance for comparing intensity	$I_{max}/32$

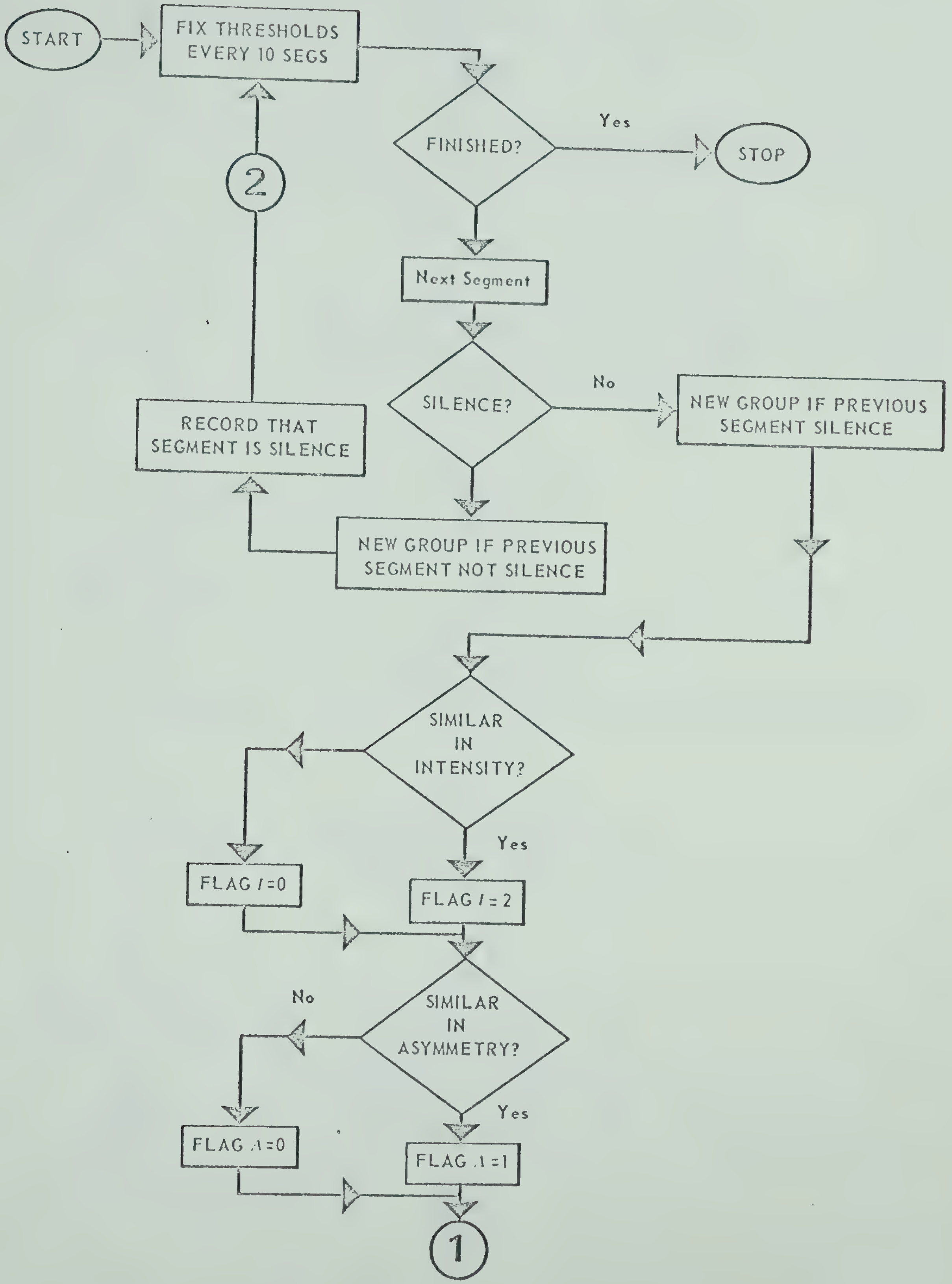

FIG. 6 FLOW-CHART FOR DECISION MAKING PROGRAM.

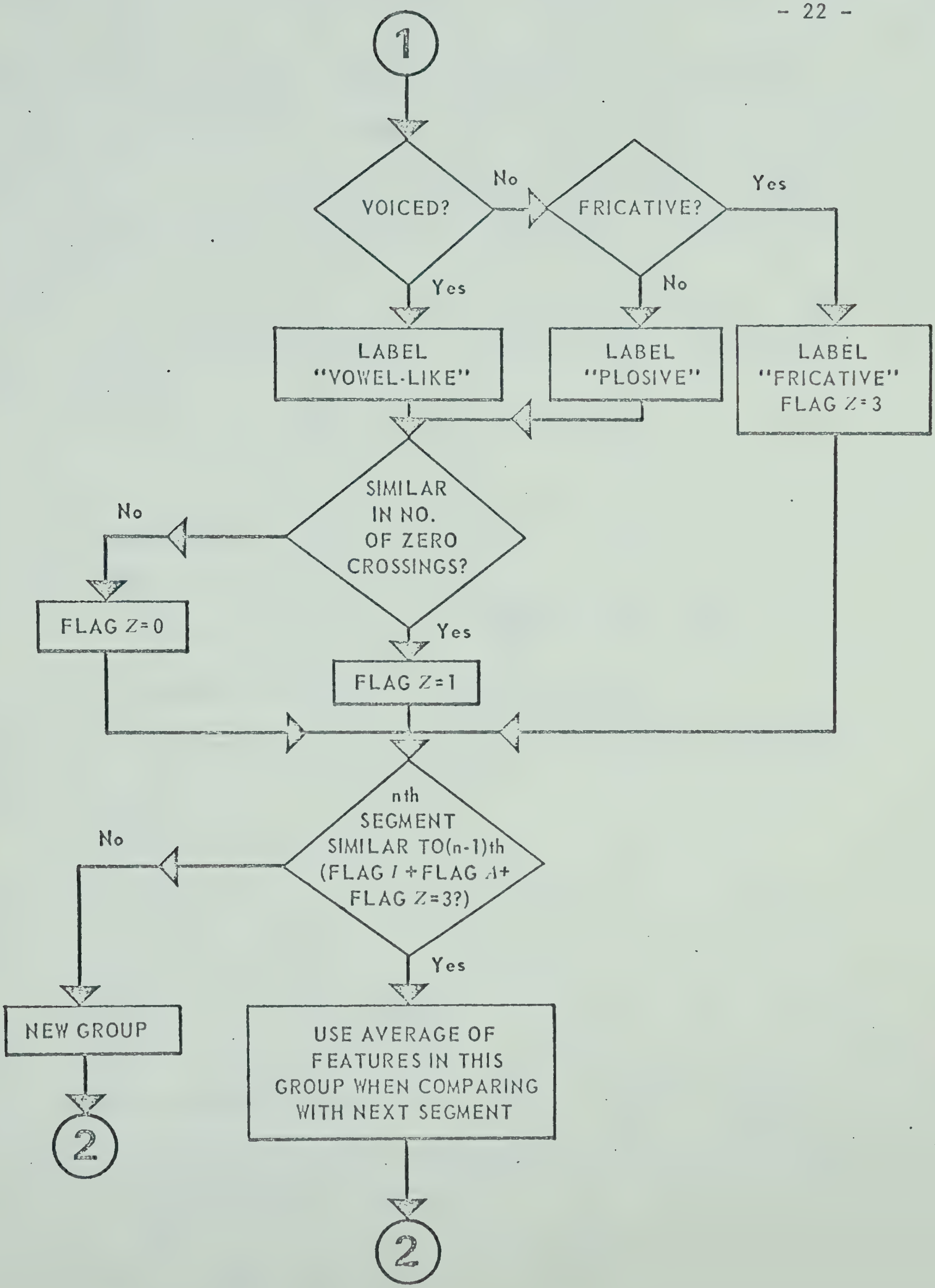
1
VOICED?
No
FRICATIVE?
Yes
Yes
No
LABEL "VOWEL-LIKE"
LABEL "PLOSIVE"
LABEL "FRICATIVE" FLAG Z=3
SIMILAR IN NO. OF ZERO CROSSINGS?
No
Yes
FLAG Z=0
FLAG Z=1
nth SEGMENT SIMILAR TO(n-1)th (FLAG I + FLAG A + FLAG Z=3?)
No
Yes
NEW GROUP
USE AVERAGE OF FEATURES IN THIS GROUP WHEN COMPARING WITH NEXT SEGMENT
2
2

FIG. 6A

are executed in that order. The three subroutines return different results to the main control program. These results depend upon whether the nth and $(n + 1)$th segments are similar or unsimilar with respect to the particular feature being compared. The similarity measures for the three features are listed below:

1. The $(n + 1)$th segment is said to be similar in "sound intensity" to the nth segment if

$$I_n - t \leq I_{n+1} \leq I_n + t$$

or

$$I_n - t \leq I_{n+2} \leq I_n + t$$

where the tolerance

$$t = \frac{I_n}{8} \text{ if } \frac{I_n}{8} \geq \frac{I_{max}}{32}, \text{ and } t = \frac{I_{max}}{32} \text{ if } \frac{I_n}{8} < \frac{I_{max}}{32}$$

I_{max} being the maximum intensity in the next 410 msecs.

2. The $(n + 1)$th segment is said to be similar in "waveform asymmetry" to the nth segment if

$A_n \varepsilon P$ and $A_{n+1} \varepsilon P$ where $P = \{A | A > W\}$

or if $A_n \varepsilon S$ and $A_{n+1} \varepsilon S$ where $S = \{A | -W \leq A \leq W\}$

or if $A_n \varepsilon N$ and $A_{n+1} \varepsilon N$ where $N = \{A | A < -W\}$

or if $A_n \varepsilon P$ and $A_{n+1} \varepsilon S$ and $A_{n+2} \varepsilon P$

or if $A_n \varepsilon N$ and $A_{n+1} \varepsilon S$ and $A_{n+2} \varepsilon N$

Here 'W' is the threshold for waveform asymmetry listed in Table 1.

3. The $(n + 1)$th segment is said to be similar in "number of zero crossings" to the nth segment if

$$Z_n - \frac{Z_n}{4} \leq Z_{n+1} \leq Z_n + \frac{Z_n}{4}$$

or

$$Z_n - \frac{Z_n}{4} \leq Z_{n+2} \leq Z_n + \frac{Z_n}{4}$$

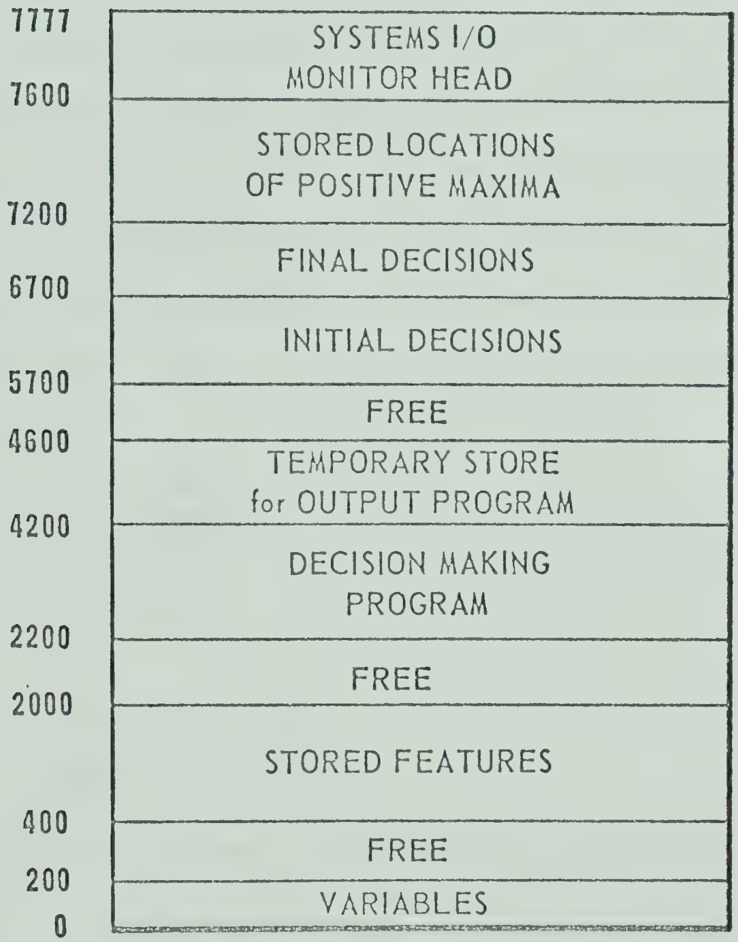

FIG. 7 CORE USAGE DIAGRAM FOR DECISION MAKING PROGRAM

where $\frac{Z_n}{4}$ is replaced by 1 if $\frac{Z_n}{4} < 1$.

In all the three similarity measures the $(n + 2)$th segment is compared to the nth if similarity between the nth and $(n + 1)$th is not found. This is to take care of the circumstance, though unlikely to occur with the segment duration of 12.8 msecs, where the segment lies entirely between two peaks of the speech wave.

When there are gradual variations in the characteristics in one direction an effective reduction in tolerance is needed to avoid grouping transitional segments together. This is obtained by using the average of the characteristics of all the segments already in the group in place of the characteristics of the nth segment in the tests for similarity. The average is found in the following manner:

$$AVG_1 = C_1$$

$$AVG_2 = \frac{AVG_1 + C_2}{2}$$

$$AVG_3 = \frac{AVG_2 + C_3}{2}$$

$$\cdot$$

$$AVG_p = \frac{AVG_{p-1} + C_p}{2}$$

where C_p is the particular characteristic for the pth segment in the group.

The main control program groups the nth and $(n + 1)$th segments together if

$$FLAGI + FLAGA + FLAGZ \geq 3$$

where FLAG*I*, FLAG*A* and FLAGZ are the output parameters of the subroutines for comparing "intensity", "asymmetry" and "zero crossings", respectively. The values given to these parameters are:

FLAGI = 2 if the segments are similar in "intensity",

and FLAGI = 0 otherwise;

FLAGA = 1 if the segments are similar in "asymmetry",

and FLAGA = 0 otherwise;

FLAGZ = 3 if the segments are grouped together as "fricative",

FLAGZ = 1 if the segments are similar in "zero crossings" but are not "fricative",

and FLAGZ = 0 otherwise.

It may be noticed that the "sound intensity" characteristic is generally given more weight. An exception to this rule, however, is made when the segments are found to be "fricative". These are noise-like segments so that both intensity and the number of zero crossings may vary considerably from segment to segment and are not reliable in finding similarity. Therefore when two segments are found to be "fricative", as defined later, the segments are always grouped together by putting FLAGZ = 3.

As similar segments are grouped together they are also tagged as "vowel-like", "silence", "fricative" or "plosive." A group is tagged "vowel-like" if the "sound intensity" of a segment in the group

$I_n \geq$ voiced-unvoiced threshold (Table 1);

or $A_n \varepsilon\ P \cup N$

"silence" if it fulfils the conditions in the test for silence described earlier;

"fricative" if the sound intensity of the nth segment

$I_n <$ voiced-unvoiced threshold, and "waveform asymmetry"

$A_n \varepsilon\ S$

and if the number of zero crossings

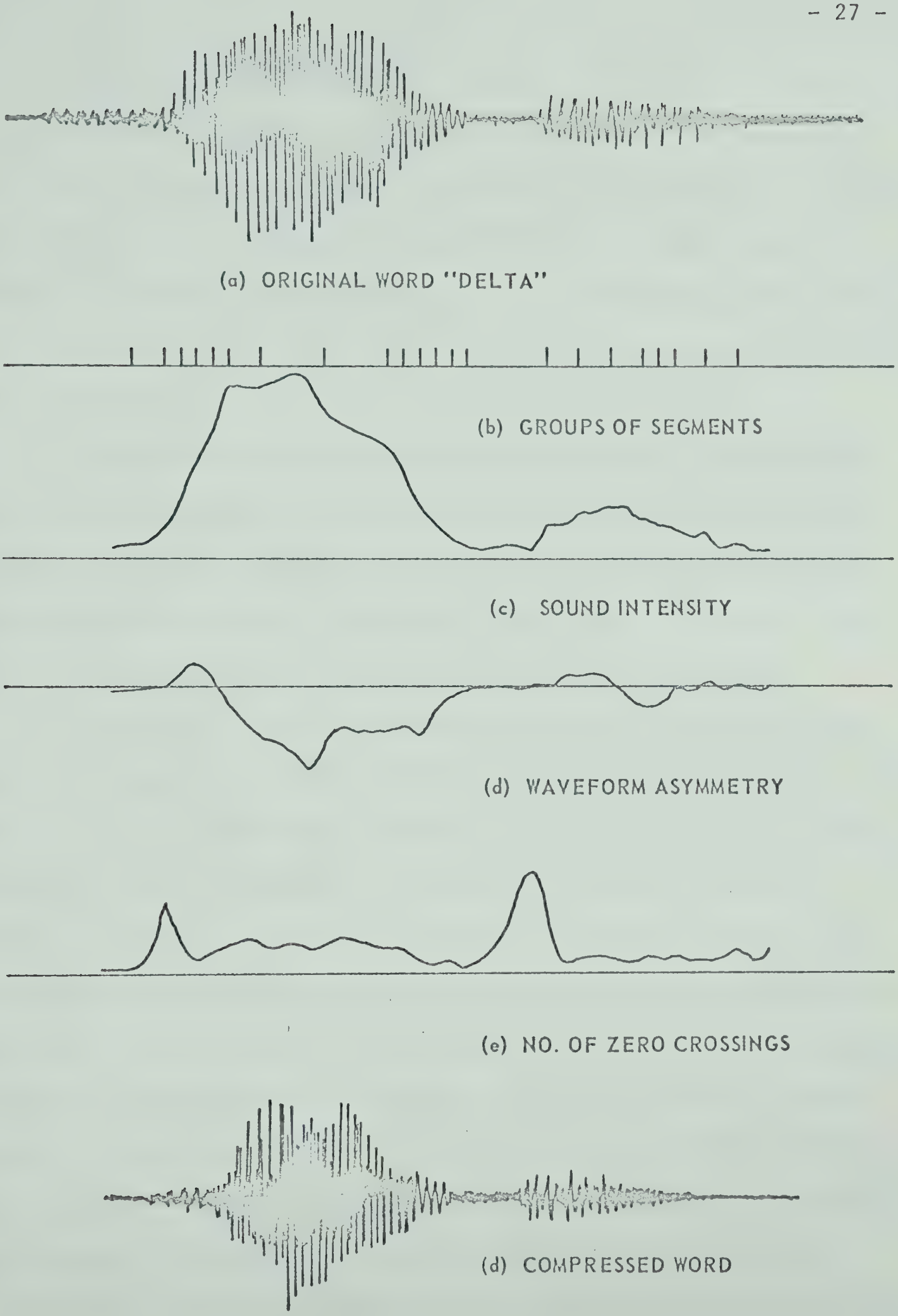

FIG. 8 ORIGINAL WORD, EXTRACTED FEATURES, AND COMPRESSED WORD

$Z_n \geq 45$ and $Z_{n+1} \geq 30$;

"plosive" if it is not any of the cases listed above. It should be noted that the classes of sounds are defined by measurements of certain characteristics of the speech waveform and not by phonetic rules. Although a realistic classification has been attempted the terms "vowel-like", "fricative" and "plosive" may not strictly correspond to similar ones in phonetics.

After the initial segmentation of speech into groups of segments with similar characteristics the various groups are searched for redundant segments. The final decisions, as to which segments are to be discarded, are based on two factors: the number of 12.8 msec segments in the group, and the type of sound the group represents.

Research on speech perception indicates that only a few complete pitch periods are sufficient for the recognition of a vowel sound. The perception of consonants, fricatives and plosives, however, depends on certain acoustic cues (24) such as burst frequency, presence or absence of friction and voicing. The effect of deleting initial parts of consonant-vowel syllables on their perception have been discussed by Grimm (25). He found that the intelligibility of these syllables falls rapidly when the syllable is truncated at the initial end to commence 50 msec or less before the peak intensity of the vowel of the syllable. The perception decreases more rapidly in the case of plosive consonants than fricative consonants. A tentative scheme, based on the above considerations, has been developed for discarding segments of different types of sounds. The rules used are as follows:

1. "Vowel-like" sounds:

No. of segments in the group	Rule
less than 3	Do not discard any segment
3	Retain one segment at each end
4 or 5	Retain one segment at one end and two at the other
6, 7 or 8	Retain two segments at each end
9 or 10	Retain one, discard two, Retain three, discard the rest retaining two at the end
11 or greater	Retain two, discard two, retain three, discard the rest retaining two at the end.

2. "Fricatives" or "silence":

No. of segments in the group	Rule
less than 5	Do not discard any segment
5 to 8 inclusive	Retain two segments at each end
9, 10 or 11	Retain three segments at each end
12 or greater	Retain four segments at each end.

3. "Plosives"

Do not discard any segment.

CHAPTER IV

TRANSIENT REMOVAL AND OUTPUT

When the redundant segments to be deleted are known the problem is how to join the remaining segments together so that there are no undesirable transients.

Three types of transients may occur. First, a local transient, that is, a discontinuity at the junction of the two retained segments. This discontinuity may be in the form of a sudden change in amplitude or an undesirable change in slope or a combination of the two. Secondly, a change in the pitch of voiced sounds. This can occur in the form of two glottal pulses lying too close or too far apart compared with the regular pitch period. Thirdly, a change in the intensity of the sound. Although a large difference between the intensities of the two segments being joined is unlikely, some amplitude smoothing in the vicinity of the junction is necessary.

The scheme for transient removal used here attempts to minimize all the three types of irregularities listed above. As mentioned earlier the location of the positive maximum in each segment is found and stored during input. Joining the segments at these points of zero slope will prevent any unwanted change in the slope of the waveform at the junction. The positive peaks of the speech waveform are also, in most cases, a fairly good approximation to the locations of

the glottal pulses in voiced sounds. Therefore joining the segments at their positive peaks will also minimize any change in pitch that may otherwise occur. This rule when combined with a rule for smoothing the amplitude at and in the vicinity of the junction will yield the desired result.

Thus if segments n to m $(n < m)$ inclusive are to be deleted segment $n - 1$ is joined with segment $m + 1$ in the following manner:

1. Let the positive maximum sample in segment $n - 1$ be the pth sample and in segment $m + 1$ be the qth sample. Then the pth sample is followed by the $(q + 1)$th sample, i.e. samples $p + 1$ to q are discarded.

2. Let the intensity of segment $n - 1$, as defined in chapter I, be I_{n-1} and of segment $m + 1$ be I_{m+1}.

   a) If $I_{n-1} > I_{m+1}$ then all the samples from $p - 63$ to p are multiplied by G where G decreases linearly in value from 1 at sample $p - 63$ to approximately

   $$I_{m+1}/I_{n-1}$$

   at sample p. The approximation is due to the fact that the slope of G is taken to be

   $$0 \text{ per sample if } \quad 0 \leq \frac{I_{n-1} - I_{m-1}}{64\ I_{n-1}} < \frac{0.5}{2047}$$

   $$\frac{1}{2047} \text{ per sample if } \frac{0.5}{2047} \leq \frac{I_{n-1} - I_{m-1}}{64\ I_{n-1}} < \frac{1.5}{2047}$$

   and so on ($\frac{1}{2047}$ is the least positive number that can be represented on the PDP-8 in single precision, taking the highest number to be 1).

b) If $I_{m+1} > I_{n-1}$ then all the samples from $q + 1$ to $q + 64$ are multiplied by G where G increases linearly from

$$I_{n-1}/I_{m+1}$$

at the $(q + 1)$th sample to approximately 1 at sample $q + 64$. The slope of G is again approximated as in case (a) above.

The flow-chart for the transient removal and output program is shown in Fig. 9. The data required for removing transients is put in order before output begins so that it is readily accessible during output. The locations of the positive peaks in segments of the type $n - 1$ and $m + 1$ are selected and stored separately. The starting value for G and its slope are then computed for all junctions of retained segments. These values are stored sequentially together with the information whether I_{n-1} is greater or less than I_{m+1} for each case.

The diagram of core usage during output is shown in Fig. 10.

Three processes proceed simultaneously during output. They are: 1) reading the stored speech samples from disk into core, 2) removal of transients and 3) D/A conversion of the samples of compressed speech at 5 KHz. Large blocks of data are read from the disk and stored temporarily in the computer core. As the digitized samples are being converted into analog voltages the desired number of samples are skipped and amplitude smoothing is done at the junctions.

Since a number of samples are skipped during output the rate of reading speech samples from the disk must be considerably higher than the rate of D/A conversion. A low rate of 5 KHz of D/A conversion has been necessitated by the limit of data transfer rate from disk to core. This is a hardware limitation. Advantage is, however, taken of this

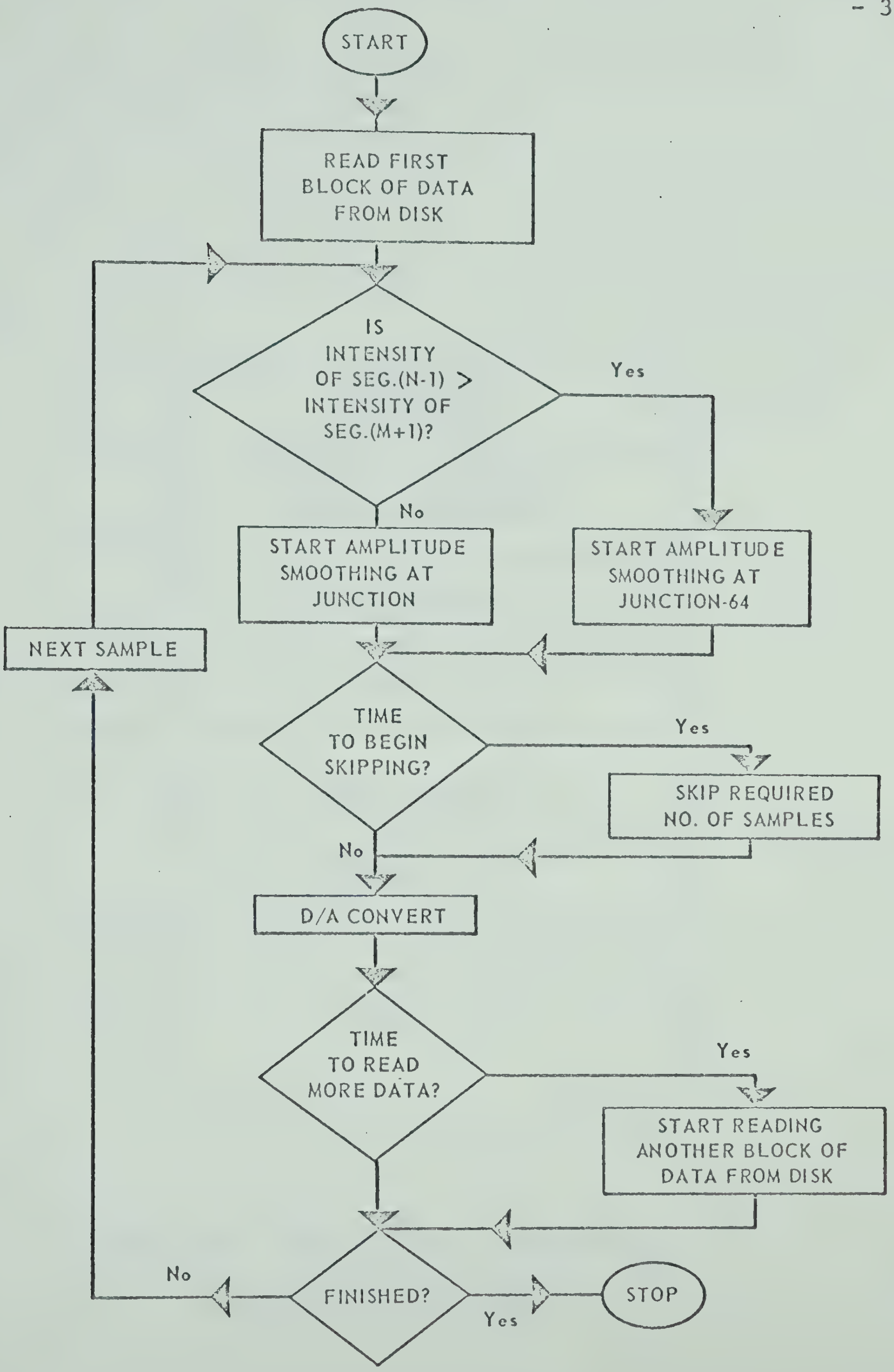

FIG. 9 FLOW-CHART OF TRANSIENT REMOVAL AND OUTPUT PROGRAM.

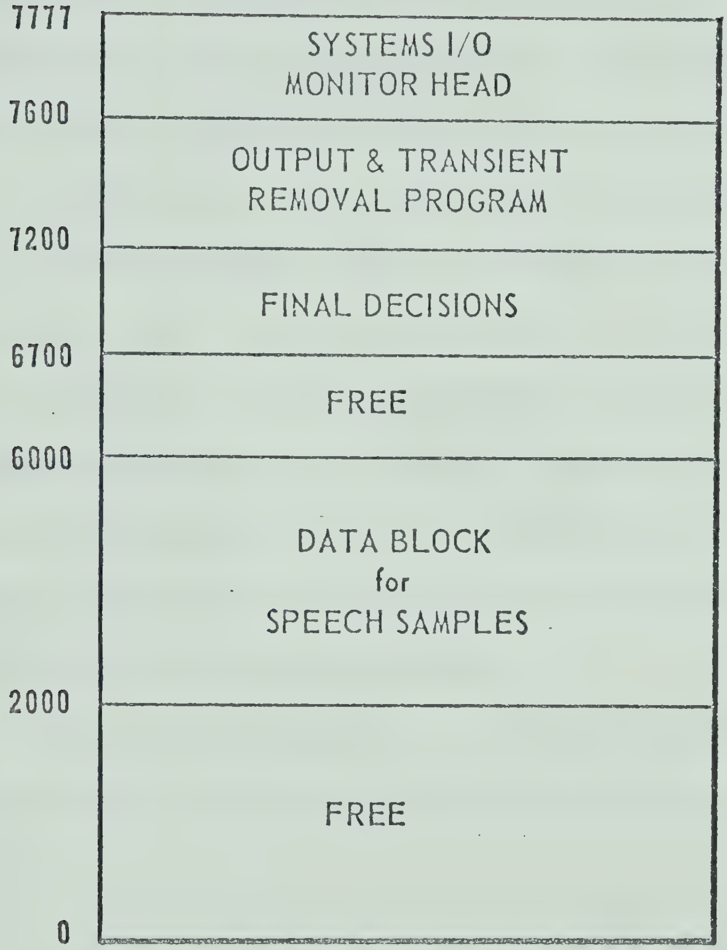

FIG. 10 CORE USAGE DIAGRAM FOR OUTPUT PROGRAM

fact and the available time is utilized in removing transients. The amount of compression obtainable is restricted because at large compressions the D/A conversion tends to go faster than the data transfer rate from disk.

An alternative scheme would be to write the speech samples back on disk after compression and transient removal. The samples could then be read again into core and D/A converted at 10 KHz to give compressed speech. This scheme would require more computer time but there is no restriction on the amount of compression.

A zero order hold circuit is present at the output of the D/A converter. This circuit holds the level of the converted analog voltage constant until the A/D conversion of the next speech sample. The stair-case waveform thus obtained is smoothed by passing it through a band-pass filter with a pass band from 40 Hz to 2.5 KHz. This speech is recorded on an ordinary tape recorder and played back at twice the speed to restore the frequency spectrum.

All the source programs are listed in Appendix B including a program for the alternative scheme for output mentioned above.

CHAPTER V

CONCLUSIONS

A selective method of speech compression has been developed by taking full advantage of the capabilities of a small inexpensive computer. It has been domonstrated that a computer with a small capacity main storage and a backing store can be used as a practical speech compressor.

Selective or differential compression has distinct advantages over a method in which portions of speech are discarded periodically. In the selective method reported here there is very little probability that portions of speech essential to its intelligibility will be removed. Compression is achieved by shortening the long inter-phrase pauses and discarding fractions of those speech sounds which have signal redundancy. Different types of speech sounds are thus treated separately.

On the present system with 4K words of core capacity and a disk memory of 32K words only short sentences have been compressed. The method, however, is attractive for compression of continuous spoken passages if a large capacity high speed disk or tape unit is connected to the computer.

The compression scheme used attempts to minimize processing time on the computer. The time taken by the decision making program is only about one fourth the original duration of the speech signal to

be compressed. This is in contrast to other methods of compressing speech by a digital computer, such as, compression by synthesis (10) or pitch period compression (15).

Apart from being a practical possibility, the method of speech compression reported here is useful as a research tool. It can be used for evaluating the effect of several factors on the intelligibility of compressed speech by varying these factors independently. It can also be used for arriving at a selective rule for compression which produces optimum intelligibility for a particular word per minute rate of speech.

A tentative rule for compression, described in chapter III, was chosen and the intelligibility of the compressed material was tested. Short sentences were compressed to an average of 70% of their original duration. The sentences used were from the Harvard Psycho-Acoustic Laboratory (P.A.L.) Auditory Test No. 12 (26). Forty-two sentences were used for the test, the first two of which are:

1. How many pennies are there in a nickel?

2. Is there a lot of water in the desert?

These questions were read by a Canadian male speaker and were recorded on tape. Subjects chosen for the listening test were of Canadian origin and had never been exposed to compressed speech. They were five undergraduate university students, four male and one female. The compressed speech was presented to the subjects by means of a loud speaker in an ordinary room. Three additional compressed sentences were first presented with the original sentences to familiarize the subjects with compressed speech. They were then asked to write the answers to the forty-two questions which followed. Four of the subjects

answered only one question wrong whereas the fifth answered three incorrectly. Although a more comprehensive study of the intelligibility of speech compressed by this method is needed the results of this preliminary test are encouraging.

Fairbanks' sampling method of speech compression was simulated on the digital computer and compared with the method reported here. Informal listening tests showed that selectively compressed speech was clearer and free of the low frequency rumble present in Fairbanks' method.

The amount of compression, though dependent on the original speech signal and its redundancy, can be changed by varying the parameters of the program. The compression can be increased by increasing the tolerance intervals for decision making and by using a rule which discards a larger fraction of a group of similar segments. The total compression for connected discourse will generally be greater than that for individual sentences because of the presence of inter-sentence pauses.

REFERENCES

1. Emerson Foulke and Thomas G. Sticht, "A review of research on the intelligibility and comprehension of accelerated speech," *Psychological Bulletin*, Vol. 72, No. 1, pp. 50-62, 1969.

2. M.R. Schroeder, "Vocoders: analysis and synthesis of speech," *Proc. I.E.E.E.*, Vol. 54, pp. 720-734, May 1966.

3. H. Fletcher, *Speech and Hearing*, New York: D. Van Nonstrand Co., pp. 293-294, 1929.

4. G.A. Miller and J.C.R. Licklider, "The intelligibility of interrupted speech," *J. Acoust. Soc. Am.*, Vol. 22, pp. 167-173, 1950.

5. W.D. Garvey, "The intelligibility of speeded speech," *J. Expt. Psychol.*, Vol. 45, No. 2, pp. 102-108, 1953.

6. Grant Fairbanks, W.L. Everitt and R.P. Jaeger, "Method for time or frequency compression-expansion of speech," *Trans. of the I.R.E. Profess. Group on Audio*, Vol. AU 2, pp. 7-12, 1954.

7. D. Gabor, "Theory of Communication," *J. Inst. Elec. Eng.*, Vol. 93, Part III, pp. 429-457, 1946.

8. D. Gabor, "New possibilities in speech transmission," *J. Inst. Elec. Eng.*, Vol. 94, Part III, pp. 369-387, 1947.

9. M.R. Schroeder, B.F. Logan and A.J. Prestigiacomo, "New methods for speech analysis-synthesis and bandwidth compression," *Congress Report I of the Fourth International Congress on Acoustics*, G41, 1962.

10. S.J. Campanella, "Signal analysis of speech time-compression techniques," *Proc. of the Louisville Conference on Time Compressed Speech* (Oct. 1966), CRCR, University of Louisville, pp. 108-113, May 1967.

11. Robert J. Scott, "Time adjustment in speech synthesis," *J. Acoust. Soc. Am.*, Vol. 41, No. 1, pp. 60-65, 1967.

12. Robert J. Scott, "Computers for speech time compression," *Proc. of the Louisville Conference on Time Compressed Speech* (Oct. 1966), CRCR, University of Louisville, pp. 29-35, May 1967.

13. S.E. Gerber, "Dichotic and diotic presentation of speeded speech," *The Journal of Communication*, Vol. 18, No. 3, pp. 272-282, 1968.

14. Emerson Foulke and E. McLean Wirth, "A comparison of 'dichotic' speech and speech compressed by the electromechanical sampling method," *The Comprehension of Rapid Speech by the Blind: Part III*, Final Progress Report (Proj. No. 2430), Non-Visual Perceptual Systems Lab., University of Louisville, pp. 30-40, September 1969.

15. W.D. Chapman, "Speech compression by tape loop and by computer," *Proc. of the Louisville Conference on Time Compressed Speech* (Oct. 1966), CRCR, University of Louisville, pp. 98-107, May 1967.

16. Emerson Foulke, "Methods for controlling the word rate of recorded speech," *The Comprehension of Rapid Speech by the Blind: Part III*, Final Progress Report (Proj. No. 2430), Non-Visual Perceptual Systems Lab., University of Louisville, pp. 21-29, September 1969.

17. P.N. Scholtz and R. Bakis, "Spoken digit recognition using vowel-consonant segmentation," *J. Acoust. Soc. Am.*, Vol. 34, pp. 1-5, 1962.

18. W.D. Gilmour, "A general purpose phonemic transcriptor," *I.E.E. N.P.L. Conference on Pattern Recognition*, pp. 154-157, 1968.

19. D.R. Reddy, "Segmentation of speech sounds," *J. Acoust. Soc. Am.*, Vol. 40, pp. 307-312, 1966.

20. David J. Comer, "The use of waveform asymmetry to identify voiced sounds," *I.E.E.E. Trans. Audio and Electroacoustics*, Vol. AU-16, pp. 500-506, December 1968.

21. T. Sakai and S. Doshita, "The automatic speech recognition system for conversational sound," *I.E.E.E. Trans. Electronic Computers*, pp. 835-846, December 1963.

22. G.W. Hughes and J.F. Hemdel, "Speech analysis," *Purdue Res. Found. Tech. Rept. TR-EE65-9*, July 1965.

23. W.C. Dersch, "Shoebox - A voice responsive machine," *Datamation*, June 1962.

24. Alvin M. Liberman, "Some results of research on speech perception," *J. Acoust. Soc. Am.*, Vol. 29, No. 1, pp. 117-123, January 1957.

25. William A. Grimm, "Perception of segments of English-spoken consonant-vowel syllables," *J. Acoust. Soc. Am.*, Vol. 40, No. 6, pp. 1454-1461, 1966.

26. C.V. Hudgins, J.E. Hawkins, J.E. Karlin and S.S. Stevens, "The development of recorded auditory tests for measuring hearing loss for speech," *The Laryngoscope*, Vol. 57, pp. 57-89, 1947.

27. S.U.H. Qureshi and Y.J. Kingma, "Time compression of speech on a small computer," Proc. of the Second Louisville Conference on Rate and/or Frequency Controlled Speech (Oct. 1969), CRCR, University of Louisville, (to be published).

APPENDIX A

ENGLISH PHONEMES*

Phonetic Symbol	Key Word	Phonetic Symbol	Key Word
Simple vowels		Plosives	
I	fit	b	bad
i	feet	d	dive
ε	let	g	give
æ	bat	p	pot
ʌ	but	t	toy
ɑ	not	k	cat
ɔ	law		
U	book	Nasal consonants	
u	boot	m	may
ɜ	bird	n	now
ə̂	Bert	ŋ	sing
Complex vowels		Fricatives	
e	pain	z	zero
o	go	ʒ	vision
aU	house	V	very
aI	ice	ð	that
ɔI	boy	h	hat
IU	few	f	fat
Semivowels and liquids		θ	thing
j	you	ʃ	shed
W	we	S	sat
l	late	Affricatives	
r	rate	tʃ	church
		dʒ	judge

* N. Lindgren, "Machine recognition of human language Part I - Automatic speech recognition," IEEE Spectrum, pp. 114-136, March 1965.

APPENDIX B

```
        /INPUT AND FEATURE EXTRACTION PROGRAM

WC=7750
CA=7751

/LIST OF VARIABLES
*0
TALLY, 0
ZCRS, 0
NEW, 0              /NEW SPEECH SAMPLE
MAXN, 0             /NEGATIVE MAXIMUM
MAXP, 0             /POSITIVE MAXIMUM
FLAG, 0             /FLAG FOR SIGN OF PREVIOUS SAMPLE
TALLY1, 0
TALLY2, 0

*20
/INITIALIZING INSTRUCTIONS
        HLT CLA
        TAD CONS
        DCA MAXLOC
        TAD CNST+1
        DCA TALLY
        TAD CONS+2
        DCA 10
        DCA I 10
        ISZ TALLY
        JMP .-2
        TAD CONS+6
        DCA 10
        TAD CNST+2
        DCA TALLY1
        TAD CNST
        DCA TALLY2
        TAD CONS+1
        DCA TALLY3
        TAD CONS+3
        DCA ZCRS
        TAD CONS+4
        DCA 15
        TAD CONS+5
        DCA 11
        DCA NEW
        TAD [EADDR-1
        DCA 16
        TAD [ADDR-1
        DCA 17
        DCA FLAG
        DCA MAXP
        DCA MAXN
        TAD CONS+1
```

```
        DCA TALLY
        6121            /SKIP ON PULSE INDICATING
       JMP .-1          /START OF SPEECH SIGNAL
        6412
        6414
        6421
        6422
        JMP ADCON

/SUBROUTINE TO WRITE FIRST BLOCK OF
/DATA ON DISK
FIRST, 0
        TAD WCOUNT
        DCA WC
        TAD CONS+6
        DCA CA
        DEAL
        TAD WCOUNT
        DMAW
        JMP I FIRST

/INSTRUCTIONS FOR READING DECISION MAKING
/AND OUTPUT PROGRAMS FROM DISK
FETCH, CLA
        TAD CONS+7
        DCA WC
        TAD TABLE
        DCA CA
        DEAL
        TAD CONS+3
        DMAR
        DFSC
        JMP .-1
        DFSE
        HLT
        JMP I CONS+10

/LIST OF CONSTANTS
CONS, 7200
        -20
        377
        400
        777
        1377
        1777
        -2400
        2200
TABLE, 2177
WCOUNT, 4000
CNST, -200
        -360
        -2600

/TRACK NUMBER ON DISK
EADDR, 100
```

```
        100
        200
        200
        300
        300
        400
        400
        500
        500
        600
        600
        700
        700
/SOME MORE VARIABLES
MAXLOC, 0
TALLY3, 0

/SUBROUTINES FOR OUTPUT PROGRAM
FINISH, 0
        DFSC
        JMP .-1
        DFSE
        HLT
        JMP I FINISH
MUL, 0
        DCA .+2
        MUY
        0
        SHL
        0
        JMP I MUL

*200
/MAIN LOOP OF INPUT AND FEATURE EXTRACTION PROGRAM
ADCON, ISZ TALLY2          /END OF SEGMENT?
        JMP IN             /NO
        TAD CNST           /YES
        DCA TALLY2
        ISZ ZCRS
        ISZ MAXLOC
        TAD MAXN
        CIA
        TAD MAXP
        SMA SZA
        JMP .+5
        DCA I 15           /WAVEFORM ASYMMETRY
        TAD MAXN
        DCA I 11           /SOUND INTENSITY
        JMP .+4
        DCA I 15
        TAD MAXP
        DCA I 11
        DCA MAXP
        DCA MAXN
        ISZ TALLY
```

```
        JMP IN
        TAD (1777
        DCA 10
        TAD (-20
        DCA TALLY
IN, TAD NEW             /NEW SAMPLE
        SPA CLA         /POSITIVE?
        JMP .+7         /NO
        TAD FLAG        /YES, SIGN OF PREVIOUS SAMPLE?
        SZA CLA         /NEGATIVE?
        JMP .+11        /NO
        IAC             /YES
        DCA FLAG
        JMP .+5
        TAD FLAG
        SNA CLA
        JMP .+3
        DCA FLAG
        ISZ I ZCRS      /ANOTHER ZERO CROSSING
        ISZ TALLY1
        JMP .+13
        ISZ TALLY3
        SKP
        JMP FETCH
LOC, JMS FIRST
        TAD GET
        DCA LOC
        TAD (JMP LOC+5
        DCA LOC+1
        TAD WCOUNT
        DCA TALLY1
        6111            /SKIP ON CLOCK PULSE
        JMP .-1
        6434            /READ A-D CONVERTER INTO AC
        DCA NEW
        TAD NEW
        DCA I 10
        6412            /RESET A-D CONVERTER
        6414            /RESET A-D INPUT MX TO CHANNEL "0"
        6421            /RESET A-D CONVERT DONE FLAG
        6422            /A-D CONVERT COMMAND
        TAD NEW
        SPA
        JMP .+12
        CIA
        TAD MAXP
        SMA CLA
        JMP ADCON
        TAD NEW
        DCA MAXP        /POSITIVE MAXIMUM
        TAD 10
        DCA I MAXLOC    /LOCATION OF POSITIVE MAX.
        JMP ADCON
        TAD MAXN
        SMA CLA
```

```
        JMP .+5
        TAD NEW
        CIA
        DCA MAXN        /NEGATIVE MAXIMUM
        JMP ADCON
        TAD MAXP
        SZA CLA
        JMP ADCON
        TAD 10
        DCA I MAXLOC
        JMP ADCON

/SUBROUTINE FOR WRITING DATA ON DISK
WRITE, 0
        DFSC
        HLT
        DFSE
        HLT
        TAD WCOUNT
        DCA WC
        TAD (1777
        DCA CA
        TAD I 16
        DEAL
        CLA
        TAD I 17
        DMAW
        JMP I WRITE
GET, JMS WRITE

/TRACK ADDRESS ON DISK
ADDR, 0
        4000
        0
        4000
        0
        4000
        0
        4000
        0
        4000
        0
        4000
        0
        4000

        $


/TO SAMPLE AT 10 KHZ CHANGE CONTENTS OF LOCATION
/CNST+2 TO "-1600".
```

```
                    /DECISION MAKING PROGRAM


/LIST OF VARIABLES
*20                 /PAGE Ø
N1, Ø               /SUBSCRIPT FOR SOUND INTENSITY
N1P1, Ø
N1P2, Ø
N2, Ø               /SUBSCRIPT FOR NO. OF
N2P1, Ø             /ZERO CROSSINGS
N2P2, Ø
N3, Ø               /SUBSCRIPT FOR WAVEFORM ASYMMETRY
N3P1, Ø
N3P2, Ø
N4, Ø
N4P1, Ø
C, Ø
SILENC, Ø           /THRESHOLD FOR SILENCE
TRSD, Ø             /THRESHOLD FOR ASYMMETRY
LIM, Ø
T1, Ø               /TOLERANCE FOR INTENSITY
T2, Ø               /TOLERANCE FOR ZERO CROSSINGS
FLAGA, Ø            /OUTPUT PARAMETERS OF SUBROUTINES
FLAGI, Ø
FLAGZ, Ø
FLAGS, Ø            /FLAG FOR SILENCE
FLAGV, Ø            /FLAG FOR VOICING
ADDR, Ø
TALLY, Ø
TALLY1, Ø
TALLY2, Ø
STORE, Ø
AVGA, Ø
AVGI, Ø
AVGZ, Ø
COUNT, Ø
TEMP, Ø
MAX, Ø
IV, Ø               /VOICED-UNVOICED THRESHOLD
IPT, Ø
IMT, Ø
XPT, Ø
XMT, Ø
TEMP1, Ø
TEMP2, Ø
LOC, Ø
GET, Ø
BEGS, Ø
PICK, Ø
ONCE, Ø
CODE, Ø             /CODE FOR TYPE OF SOUND
NCODE, Ø
EPICK, Ø
ENDS, Ø
```

```
*2200
/INITIALIZING INSTRUCTIONS
INITLZ, CLA CMA             /GIVE INITIAL VALUES TO
        DCA TALLY1          /VARIABLES
        DCA FLAGS
        TAD (777
        DCA N3
        TAD (1000
        DCA N3P1
        TAD (1001
        DCA N3P2
        DCA FLAGA
        TAD (1377
        DCA N1
        TAD (1400
        DCA N1P1
        TAD (1401
        DCA N1P2
        DCA FLAGI
        TAD (377
        DCA N2
        TAD (400
        DCA N2P1
        TAD (401
        DCA N2P2
        DCA FLAGZ
        CLA CMA
        DCA ONCE
        JMP GO

/MAIN CONTROL PROGRAM
MAIN, ISZ C
        ISZ TALLY           /FINISHED?
        SKP                 /NO
        JMP 3200            /YES
        TAD (-1717
        TAD STORE
        SMA SZA CLA
        JMP .+6
        ISZ TALLY1
        SKP
        JMS FIX             /FIX THRESHOLDS EVERY
        TAD (-12            /TEN SEGMENTS
        DCA TALLY1
        ISZ N1              /INCREMENT SUBSCRIPTS
        ISZ N1P1
        ISZ N2
        ISZ N2P1
        TAD I N1            /CHECK FOR SILENCE
        TAD SILENC
        SMZ SZA CLA
        JMP OUT             /NOT SILENCE
        TAD I N1P1
        TAD SILENC
        SMA SZA CLA
```

```
        JMP OUT
        TAD I N2
        TAD (-20
        SMA SZA CLA
        JMP OUT
        TAD I N2P1
        TAD (-20
        SMA SZA CLA
        JMP OUT
        TAD FLAGS           /PREVIOUS SEGMENT SILENCE?
        SZA CLA
        JMP CONT            /YES
        ISZ ONCE
        SKP
        JMP .+16
        TAD 10              /NO, REGISTER BREAK
        DCA ADDR
        CMA
        TAD C
        CIA
        TAD I ADDR
        SNA CLA
        JMP .+6
        CMA
        TAD C
        DCA I 10
        TAD CODE
        DCA I 11
        IAC
        DCA FLAGS
        JMP CONT
OUT,  TAD FLAGS             /NOT SILENCE
        SNA CLA             /PREVIOUS SEGMENT SILENCE?
        JMP .+7             /NO
        DCA FLAGS           /YES
        TAD C               /REGISTER BREAK
        DCA I 10
        IAC                 /TAG GROUP AS SILENCE
        DCA I 11
        JMP CONT
        JMS ASYM            /EXECUTE SUBROUTINES
        JMS INTNS
        JMS ZEROX
DCIDE, TAD FLAGA            /DECIDE
        TAD FLAGI
        TAD FLAGZ
        TAD (-3
        SPA CLA             /ARE SEGMENTS SIMILAR?
        JMP BRK             /NO, REGISTER BREAK
        TAD NCODE           /YES
        DCA CODE
        JMP MAIN

*2400
/SUBROUTINE FOR COMPARING WAVEFORM ASYMMETRY
```

```
ASYM, 0
        ISZ N3              /INCREMENT SUBSCRIPTS
        ISZ N3P1
        ISZ N3P2
        TAD FLAGA           /FIND AVERAGE IF SAME GROUP
        SZA CLA
        JMP .+3
        TAD I N3
        JMP .+5
        TAD AVGA
        TAD I N3
        ASR
        0
        DCA AVGA
        TAD (-2
        DCA TALLY2
        TAD AVGA
        SPA                 /IS ASYMMETRY POSITIVE?
        JMP .+6             /NO
        CIA                 /YES
        TAD TRSD
        SMA SZA CLA         /GREATER THAN OR EQUAL TO TRSD?
        JMP TWO             /NO
        JMP ONE             /YES
        TAD TRSD
        SMA SZA CLA
        JMP TWO
        IAC
        DCA FLAGV           /SET FLAG FOR VOICING
THREE, TAD I N3P1           /WAVEFORM HAS NEGATIVE ASYMMETRY
        SPA
        JMP .+6
        CIA
        TAD TRSD
        SMA SZA CLA
        JMP .+5
        JMP BRKA
        TAD TRSD
        SPA CLA
        JMP CONTA
        TAD I N3P2
        SMA SZA
        JMP BRKA
        TAD TRSD
        SMA SZA CLA
        JMP BRKA
        JMP CONTA
TWO, DCA FLAGV              /WAVEFORM IS ALMOST SYMMETRICAL
        TAD I N3P1
        SPA
        JMP .+6
        CIA
        TAD TRSD
        SMA SZA CLA
        JMP CONTA
```

```
        JMP .+4
        TAD TRSD
        SMA SZA CLA
        JMP CONTA
        TAD I N3P2
        ISZ TALLY2
        JMP TWO+2
        JMP BRKA
ONE, CLA IAC              /WAVEFORM HAS POSITIVE ASYMMETRY
        DCA FLAGV         /SET FLAG FOR VOICING
        TAD I N3P1
        SMA SZA
        JMP .+5
        TAD TRSD
        SMA SZA CLA
        JMP .+6
        JMP BRKA
        CIA
        TAD TRSD
        SPA CLA
        JMP CONTA
        TAD I N3P2
        SPA
        JMP BRKA
        CIA
        TAD TRSD
        SMA SZA CLA
        JMP BRKA
CONTA, CLA IAC            /SIMILAR IN ASYMMETRY
        DCA FLAGA
        JMP I ASYM
BRKA, CLA                 /UNSIMILAR IN ASYMMETRY
        DCA FLAGA
        JMP I ASYM

/PARTS OF MAIN CONTROL PROGRAM
CONT, ISZ N3              /SEGMENT WAS SILENCE
        ISZ N3P1          /INCREMENT SUBSCRIPTS
        ISZ N3P2
        ISZ N1P2
        ISZ N2P2
        TAD NCODE
        DCA CODE
        JMP MAIN
BRK, TAD C                /SEGMENTS UNSIMILAR
        DCA I 10          /RECORD BREAK
        TAD CODE          /RECORD TYPE OF SOUND
        DCA I 11
        TAD NCODE
        DCA CODE
        JMP MAIN

/INITIALIZING INSTRUCTIONS
GO, TAD N1
        DCA STORE
```

```
        DCA C
        TAD (-360
        DCA TALLY
        TAD (6377
        DCA 10
        TAD (5777
        DCA 11
        DCA CODE
        JMP MAIN

/UTILITY SUBROUTINE
CHANGE, 0                       /SHIFT DATA IN CORE
        TAD I 10
        DCA I 11
        ISZ TALLY
        JMP .-3
        JMP I CHANGE

*2600
/SUBROUTINE FOR SETTING THRESHOLDS
FIX, 0
        TAD STORE
        DCA 17
        DCA MAX
        TAD (-40
        DCA COUNT
        CLA
        ISZ COUNT
        SKP
        JMP .+11
        TAD MAX                 /FIND MAXIMUM INTENSITY IN
        CIA                     /NEXT 32 SEGMENTS
        TAD I 17
        SPA
        JMP .-10
        TAD MAX
        DCA MAX
        JMP .-12
        TAD MAX
        CLL RAR
        SZL
        IAC
        CLL RAR
        SZL
        IAC
        DCA IV                  /VOICED-UNVOICED THRESHOLD
        TAD IV
        CLL RAR
        SZL
        IAC
        DCA TRSD                /THRESHOLD FOR ASYMMETRY
        TAD TRSD
        CLL RAR
        SZL
        IAC
```

```
        DCA SILENC
        TAD SILENC
        CLL RAR
        SZL
        IAC
        DCA LIM          /MININMUM TOLERANCE FOR
        TAD LIM          /COMPARING INTENSITY
        CLL RAR
        SZL
        IAC
        TAD SILENC
        CIA
        DCA SILENC       /THRESHOLD FOR SILENCE
        TAD STORE
        TAD (12
        DCA STORE
        JMP I FIX

/SUBROUTINE FOR COMPARING SOUND INTENSITY
INTNS, 0
        ISZ N1P2
        TAD FLAGI        /FIND AVERAGE IF SAME GROUP
        SZA CLA
        JMP .+3
        TAD I N1
        JMP .+6
        TAD AVGI
        TAD I N1
        CLL RAR
        SZL
        IAC
        DCA AVGI
        TAD AVGI
        CLL RAR
        SZL
        IAC
        CLL RAR
        SZL
        IAC
        CLL RAR
        SZL
        IAC
        DCA T1
        TAD LIM
        CIA
        TAD T1
        SMA CLA
        JMP .+3
        TAD LIM
        DCA T1           /TOLERANCE FOR INTENSITY
        TAD AVGI
        TAD T1
        DCA IPT          /INTENSITY PLUS TOLERANCE
        TAD T1
        CIA
```

```
        TAD AVGI
        DCA IMT         /INTENSITY MINUS TOLERANCE
        TAD I N1P1      /INTENSITY OF NEXT SEGMENT
        CIA
        TAD IPT
        SPA CLA         /LESS THAN OR EQUAL TO IPT?
        JMP .+6         /NO
        TAD IMT         /YES
        CIA
        TAD I N1P1
        SMA CLA         /GREATER THAN OR EQUAL TO IMT?
        JMP CONTI       /YES
        TAD I N1P2      /NO, CHECK SEGMENT NEXT+1
        CIA
        TAD IPT
        SPA CLA
        JMP BRKI
        TAD IMT
        CIA
        TAD I N1P2
        SPA CLA
        JMP BRKI
CONTI, CLA              /SIMILAR IN INTENSITY
        TAD (2
        DCA FLAGI
        JMP I INTNS
BRKI, DCA FLAGI         /UNSIMILAR IN INTENSITY
        JMP I INTNS

/UTILITY SUBROUTINE
LET, 0
        TAD (5670
        DCA BEGS
        TAD (-44
        DCA TALLY
        JMP I LET

*3000
/SUBROUTINE FOR COMPARING NUMBER OF ZERO
/CROSSINGS
ZEROX, 0
        ISZ N2P2
        TAD FLAGZ       /FIND AVERAGE IF SAME GROUP
        SZA CLA
        JMP .+3
        TAD I N2
        JMP .+6
        TAD AVGZ
        TAD I N2
        CLL RAR
        SZL
        IAC
        DCA AVGZ
        TAD AVGZ
        CLL RAR
```

```
SZL
IAC
CLL RAR
SZL
IAC
SNA
IAC
DCA T2          /TOLERANCE FOR ZERO CROSSINGS
TAD AVGZ
TAD T2
DCA XPT         /ZERO CROSSINGS PLUS TOLERANCE
TAD T2
CIA
TAD AVGZ
DCA XMT         /ZERO CROSSINGS MINUS TOLERANCE
TAD AVGI        /LABEL THE SEGMENT AS "VOWEL-LIKE"
CIA             /"FRICATIVE", OR "PLOSIVE"
TAD IV
SPA CLA
JMP .+21
TAD FLAGV
SZA CLA
JMP .+16
TAD AVGZ
TAD (-55
SPA CLA
JMP .+11
TAD I N2P1
TAD (-36
SPA CLA
JMP .+5
TAD (2          /"FRICATIVE"
DCA NCODE
TAD (3
JMP CONTZ+1
TAD (3          /"PLOSIVE"
DCA NCODE       /"VOWEL-LIKE"
TAD I N2P1      /# ZERO CROSSINGS IN NEXT SEGMENT
CIA
TAD XPT
SPA CLA         /LESS THAN OR EQUAL TO XPT?
JMP .+6         /NO
TAD XMT         /YES
CIA
TAD I N2P1
SMA CLA         /GREATER THAN OR EQUAL TO XMT?
JMP CONTZ       /YES
TAD I N2P2      /NO, CHECK SEGMENT NEXT+1
CIA
TAD XPT
SPA CLA
JMP BRKZ
TAD XMT
CIA
TAD I N2P1
```

```
        SPA CLA
        JMP BRKZ
CONTZ,  CLA IAC                 /SIMILAR IN NUMBER OF
        DCA FLAGZ               /ZERO CROSSINGS
        JMP I ZEROX
BRKZ,   DCA FLAGZ               /UNSIMILAR IN NUMBER OF
        JMP I ZEROX             /ZERO CROSSINGS

/PART OF FINAL DECISION MAKING PROGRAM
SIL,    CLA                     /SILENCE
        TAD I TEMP1
        CIA
        TAD I TEMP2
        TAD (-5
        SPA
        JMP IN
        TAD (-4
        SPA
        JMP DO
        TAD (-3
        SPA CLA
        JMP .+10
        TAD (4
        TAD I TEMP1
        DCA I 16
        TAD (-4
        TAD I TEMP2
        DCA I 17
        JMP IN
        TAD (3
        TAD I TEMP1
        DCA I 16
        TAD (-3
        JMP DO+5
DO,     CLA
        TAD (2
        TAD I TEMP1
        DCA I 16
        TAD (-2
        TAD I TEMP2
        DCA I 17
        JMP IN

*3200
/FINAL DECISION MAKING PROGRAM
FINAL,  CLA
        TAD (5777               /INITIALIZE
        DCA 10
        TAD (-110
        DCA TALLY
        TAD (5667
        DCA 16
        DCA I 16
        ISZ TALLY
        JMP .-2
```

```
        TAD (5667
        DCA 16
        TAD (5733
        DCA 17
        TAD (6400
        DCA N4
        TAD (6401
        DCA N4P1
        TAD (-200
        DCA TALLY
        TAD N4
        DCA TEMP1
        TAD I 10        /DECODE TYPE OF SOUND
        SNA             /FOR FIRST GROUP
        JMP VWL         /"VOWEL-LIKE"
        TAD (-2
        SZA CLA
        JMP RING+5      /"PLOSIVE", NO DELETIONS
        TAD I TEMP1     /"FRICATIVE"
        TAD (-5
        SPA
        JMP RING+2
        TAD (-3
        SPA
        JMP .+16
        TAD (-3
        SPA CLA
        JMP .+7
        TAD (4
        DCA I 16
        TAD (-4
        TAD I TEMP1
        DCA I 17
        JMP RING+5
        TAD (3
        DCA I 16
        TAD (-3
        JMP .-6
        CLA
        TAD (2
        DCA I 16
        TAD (-2
        JMP .-13
VWL,    CLA             /"VOWEL-LIKE"
        TAD I TEMP1     /#SEGMENTS IN FIRST GROUP
        TAD (-4
        SPA             /GREATER THAN OR EQUAL TO 4
        JMP RING+2      /NO, DELETE NOTHING
        TAD (-2         /YES
        SPA
        JMP .+22
        TAD (-3
        SMA CLA
        JMP .+6
        TAD (2
```

```
        DCA I 16
        TAD (4
        DCA I 17
        JMP RING+5
        TAD (2
        DCA I 16
        TAD (4
        DCA I 17
        TAD (6
        DCA I 16
        TAD (10
        DCA I 17
        JMP RING+5
        CLA
        TAD (2
        DCA I 16
        TAD (-1
        TAD I TEMP1
        DCA I 17
        JMP RING+5
REM, CLA IAC
        TAD I TEMP1
        DCA I 16
        TAD (-1
        TAD I TEMP2
        DCA I 17
        JMP IN

/INSTRUCTIONS FOR TRANSFERRING CONTROL TO
/OUTPUT PROGRAM
SLEEP, TAD (4177
        DCA 10
        TAD (7177
        DCA 11
        TAD (-400
        DCA TALLY
        JMS CHANGE
        JMP 7200

*3400
/FINAL DECISION MAKING PROGRAM CONTINUED
RING, ISZ N4
        ISZ N4P1
        CLA
        TAD N4
        DCA TEMP1
        TAD N4P1
        DCA TEMP2
        TAD I TEMP2
        SMA SZA CLA
        SKP
        JMP IN+2
        TAD I 10        /DECODE TYPE OF SOUND FOR
        SNA             /ALL GROUPS AFTER FIRST
        JMP VOWL        /"VOWEL-LIKE"
```

```
        TAD (-1
        SNA
        JMP SIL             /SILENCE
        TAD (-1
        SZA CLA
        JMP IN              /"PLOSIVE", NO DELETIONS
        JMP SIL             /"FRICATIVE"
VOWL, CLA                   /"VOWEL-LIKE"
        TAD I TEMP1
        CIA
        TAD I TEMP2
        TAD (-3
        SPA
        JMP IN
        TAD (-1
        SPA
        JMP REM
        TAD (-2
        SPA
        JMP DEL
        TAD (-3
        SPA
        JMP DO
        TAD (-2
        SPA CLA
        JMP CEL
        TAD (2
        TAD I TEMP1
        DCA I 16
        TAD (4
        TAD I TEMP1
        DCA I 17
        TAD (7
        TAD I TEMP1
        DCA I 16
        JMP DEL+3
CEL, IAC
        TAD I TEMP1
        DCA I 16
        TAD (3
        TAD I TEMP1
        DCA I 17
        TAD (6
        TAD I TEMP1
        DCA I 16
        JMP DEL+3
DEL, CLA IAC
        TAD I TEMP1
        DCA I 16
        TAD (-2
        TAD I TEMP2
        DCA I 17
IN, ISZ TALLY
        JMP RING
        CLA
```

```
        TAD (6677          /INITIALIZATION OF FIRST PART
        DCA 15             /OF TRANSIENT REMOVAL PROGRAM
        TAD (-200
        DCA TALLY
        DCA I 15
        ISZ TALLY
        JMP .-2
        TAD (6757
        DCA 15
        TAD (7177
        DCA ADDR
        JMS LET
        TAD (6677
        DCA 16
        IAC
        DCA TALLY1
        JMP START

*3600
/FIRST PART OF TRANSIENT REMOVAL PROGRAM
BEGIN, TAD (JMP NOW
        DCA A+6
START, TAD ADDR
        TAD I BEGS
        DCA PICK
        TAD I PICK
        DCA I 15           /LOCATIONS OF POSITIVE PEAKS TO
B, NOP                     /BE USED DURING TRANSIENT REMOVAL
        TAD I BEGS
        TAD VALUE
        SPA
        JMP FIRST
        TAD (-20
        SPA
        JMP FAST
        ISZ TALLY1
        JMP .-4
FIRST, CLA
        TAD (17
        DCA I 16
        JMP A
FAST, CLA
        TAD TALLY1
        CIA
        TAD (17
        DCA I 16
        IAC
        DCA TALLY1
A, ISZ BEGS
        TAD I BEGS
        SPA SNA CLA
        JMP .+3
        ISZ TALLY
        JMP START
        JMP AGAIN
```

```
AGAIN, TAD (7200
        DCA ADDR
        TAD (5734
        DCA BEGS
        TAD (7023
        DCA 15
        TAD (JMP A
        DCA B
        TAD (-44
        DCA TALLY
        JMP BEGIN

/INITIALIZING INSTRUCTIONS FOR AMPLITUDE SMOOTING
NOW, TAD (1377
        DCA ADDR
        JMS LET
        TAD (5734
        DCA ENDS
        TAD (7067
        DCA 14
        TAD (7133
        DCA 15
        JMP BACK

        TAD (3777       /STARTING VALUE OF GAIN "G"
        DCA I 14
FISH, ISZ BEGS          /INCREMENT SUBSCRIPTS
        ISZ ENDS
        TAD I BEGS
        SPA SNA CLA
        JMP .+3
        ISZ TALLY       /FINISHED?
        JMP BACK        /NO
        JMP SLEEP       /YES, READ OUTPUT PROGRAM
VALUE, -14

*4000
/FIRST PART OF TRANSIENT REMOVAL PROGRAM
/INSTRUCTIONS FOR AMPLITUDE SMOOTHING
BACK, IAC
        TAD ADDR
        TAD I ENDS
        DCA EPICK
        TAD ADDR
        TAD I BEGS
        DCA PICK
        TAD I PICK      /INTENSITY OF SEGMENT (N-1)
        CIA
        TAD I EPICK     /INTENSITY OF SEGMENT (M+1)
        SPA CLA         /(WHERE SEGMENTS N TO M ARE
        JMP SMOKE       /TO BE DISCARDED)
        MQL
        TAD I EPICK     /INTENSITY OF (M+1) IS GREATER
        DCA .+5         /THAN THAT OF (N-1)
        TAD I PICK
```

```
        LSR
        Ø
        DVI
        Ø
        CLA MQA
        DCA TEMP
        TAD TEMP
        CIA
        TAD (3737
        SPA
        JMP .+40
        TAD (-100
        SPA
        JMP .+33
        TAD (-100
        SPA
        JMP .+25
        TAD (-100
        SPA
        JMP .+17
        TAD (-100
        SPA
        JMP .+11
        TAD (-100
        SPA CLA
        JMP .+4
        TAD (6
        DCA I 15        /SLOPE OF "G" PER SAMPLE
        JMP .+20
        TAD (5
        JMP .-3
        CLA
        TAD (4
        JMP .-6
        CLA
        TAD (3
        JMP .-11
        CLA
        TAD (2
        JMP .-14
        CLA IAC
        JMP .-16
        CLA
        JMP .-20
        TAD TEMP
        DCA I 14        /STARTING VALUE OF GAIN "G"
        JMP FISH
SMOKE,  MQL             /INTENSITY OF (N-1) IS GREATER
        TAD I PICK      /THAN THAT OF (M+1)
        DCA .+5
        TAD I EPICK
        LSR
        Ø
        DVI
        Ø
```

```
        CLA MQA
        CIA
        TAD (3737
        SPA
        JMP .+40
        TAD (-100
        SPA
        JMP .+33
        TAD (-100
        SPA
        JMP .+25
        TAD (-100
        SPA
        JMP .+17
        TAD (-100
        SPA
        JMP .+11
        TAD (-100
        SPA CLA
        JMP .+4
        TAD (-6
        DCA I 15         /SLOPE OF "G" PER SAMPLE
        JMP FISH-2
        TAD (-5
        JMP .-3
        CLA
        TAD (-4
        JMP .-6
        CLA
        TAD (-3
        JMP .-11
        CLA
        TAD (-2
        JMP .-14
        CLA CMA
        JMP .-16
        CLA
        JMP .-20

        $


/       THE INPUT PROGRAM IS STORED ON TRACK ZERO OF
/THE DISK FROM LOCATION 0 TO 377. THE OUTPUT PROGRAM
/IS FIRST SHIFTED IN CORE FROM 7200 - 7577 TO 4200 -
/4577. THE DECISION MAKING AND OUTPUT PROGRAMS ARE
/THEN STORED ON TRACK ZERO OF THE DISK FROM LOCATION
/400 TO 3000. SEPARATE PROGRAMS WERE WRITTEN FOR THIS
/PURPOSE AND FOR READING THE INPUT PROGRAM INTO CORE.
```

```
        /TRANSIENT REMOVAL AND OUTPUT PROGRAM  1

WC=7750
CA=7751

/LIST OF VARIABLES
*20
GET, 0
FLAG, 0
FLAG1, 0
TALLY, 0
TALLY2, 0
ENDS, 0
BEGS, 0
LIMP, 0
LIMN, 0

*7200
/INSTRUCTIONS FOR READING FIRST
/BLOCK OF DATA FROM DISK
        HLT CLA
        TAD CONS+4
        DCA WC
        TAD CONS+5
        DCA CA
        DEAL
        TAD CONS+4
        DMAR

/INITIALIZING INSTRUCTIONS
        TAD CONS
        DCA BEGS
        TAD CONS+1
        DCA ENDS
        TAD CONS+2
        DCA TALLY2
        TAD CONS+3
        DCA TALLY
        TAD CONS+5
        DCA 10
        CLA IAC
        DCA FLAG
        DCA FLAG1
        TAD CONS+6
        DCA LIMP
        TAD CONS+7
        DCA LIMN
        TAD (6700
        DCA GET
        TAD (EADDR-1
        DCA 11
        TAD (ADDR-1
        DCA 12
        DFSC
        JMP .-1
```

```
CHECK, TAD I GET
        TAD TALLY
        SNA CLA
        JMP .+4
        TAD OFF1
        DCA DACON
        JMP DACON
        ISZ GET
        JMP DACON

/LIST OF CONSTANTS
CONS, 6760
        7024
        -101
        -17
         4000
        1777
        7070
        7134


/PART OF MAIN LOOP OF OUTPUT PROGRAM
        TAD CONS+2
        DCA TALLY2
        ISZ LIMN
        ISZ LIMP
        DCA FLAG1
        JMP OUT+2
OUT, TAD I 10
        6464                /LOAD DATA OUTPUT BUFFER
        6452                /RESET CHANNEL ADDRESS
        6111                /SKIP ON CLOCK PULSE
        JMP .-1
        6454                /D-A CONVERT
        CLA
        JMP DACON
/TRACK NUMBER ON DISK
EADDR, 100
        100
        200
        200
        300
        300
        400
        400
        500
        500
        600
        600
        700
        700

/TRACK ADDRESS ON DISK
ADDR, 0
        4000
```

```
        0
        4000
        0
        4000
        0
        4000
        0
        4000
        0
        4000
        0
        4000

*7400
/MAIN LOOP OF TRANSIENT REMOVAL AND
/OUTPUT PROGRAM
DACON,  NOP
        NOP
        TAD I LIMN
        SMA CLA
        JMP .+13
        TAD 10
        SMA
        JMP .+6
        TAD (-5677
        SPA CLA
        JMP .+4
        TAD (4100
        JMP .+3
        CLA
        TAD (100
        TAD 10
        CIA
        TAD I BEGS
        SZA CLA
        JMP NOW
        TAD OFF
        DCA DACON+1
        IAC
        DCA FLAG1
NOW, TAD 10
        CIA
        TAD I BEGS
        SZA CLA
        JMP .+16
        TAD I ENDS
        DCA 10
LET, TAD I GET
        TAD TALLY
        SZA CLA
        JMP .+6
        ISZ GET
        ISZ BEGS
        ISZ ENDS
        TAD ON
```

```
        SKP
        TAD OFF1
        DCA DACON
        TAD 10
        TAD (-5777
        SZA CLA
        JMP .+6
        TAD CONS+5
        DCA 10
LOC,    CLA IAC
        DCA FLAG
        JMP IN
        TAD 10
        SMA
        JMP LOC
        TAD (-4510
        SPA CLA
        JMP LOC
        TAD FLAG
        SNA CLA
        JMP IN
        DCA FLAG
        ISZ TALLY
        JMP .+5
        6452            /RESET CHANNEL ADDRESS
        6462
        6454            /D-A CONVERT
        HLT
        DFSC
        HLT
        DFSE
        HLT
        TAD WCOUNT
        DCA WC
        TAD CONS+5
        DCA CA
        TAD I 11
        DEAL
        CLA
        TAD I 12
        DMAR

/INSTRUCTIONS FOR AMPLITUDE SMOOTHING
IN,     6462            /RESET DATA AND OUTPUT BUFFERS
        TAD FLAG1
        SNA CLA
        JMP OUT
        TAD I LIMP
        TAD I LIMN
        DCA I LIMP
        TAD I LIMP
        MQL
        TAD I 10
        SPA
        JMP .+7
```

```
        DCA .+2
        MUY
        Ø
        SHL
        Ø
        JMP DRINK-1
        CIA
        DCA .+2
        MUY
        Ø
        SHL
        Ø
        CIA
        6464            /LOAD DATA OUTPUT BUFFER
DRINK, ISZ TALLY2
        JMP OUT+2
        CLA
        TAD ON
        DCA DACON+1
        JMP OUT-6
ON, NOP
OFF, JMP NOW
OFF1, JMP LET
WCOUNT, 4ØØØ

        $
```

```
        /TRANSIENT REMOVAL AND OUTPUT PROGRAM II

WC=7750
CA=7751
WCOUNT=131
FINISH=155
MUL=163

/LIST OF VARIABLES
*20
TALLY, 0
TALLY1, 0
BEGS, 0
ENDS, 0
FLAG, 0
FLAG1, 0
LIMP, 0
LIMN, 0
WEAD, 0
WADD, 0
GET, 0
ADDR, 0
EADDR, 0
FLAG2, 0
COUNT, 0

*7200
/INITIALIZING INSTRUCTIONS
        TAD (1777
        DCA 10
        DCA EADDR
        DCA ADDR
        JMS DO
        TAD (6760
        DCA BEGS
        TAD (7024
        DCA ENDS
        TAD (-101
        DCA TALLY1
        TAD WCOUNT
        DCA WADD
        DCA FLAG1
        DCA FLAG2
        TAD (7070
        DCA LIMP
        TAD (7134
        DCA LIMN
        TAD (6700
        DCA GET
        DCA WEAD
        TAD (-17
        DCA TALLY
        TAD (-17
        DCA COUNT
CHECK, TAD I GET
```

```
        TAD COUNT
        JMP 7400

/SUBROUTINE FOR READING DATA FROM DISK
READ, 0
        TAD WCOUNT
        DCA WC
        TAD (1777
        DCA CA
        TAD EADDR
        TAD (40
        DCA EADDR
        TAD EADDR
        AND (700
        DEAL
        CLA
        TAD ADDR
        TAD WCOUNT
        DCA ADDR
        TAD ADDR
        DMAR
        JMP I READ
DO, 0
        JMS READ
        CLA IAC
        DCA FLAG
        TAD (1777
        DCA 11
        JMP I DO

/PART OF THE MAIN PROGRAM LOOP FOR COMPRESSION
BACK, TAD WADD
        DMAW
        JMS FINISH
        DMAC
        CLL IAC
        DCA WADD
        DEAC
        NOP
        SZL
        TAD (100
        DCA WEAD
        ISZ TALLY       /FINISHED?
        SKP             /NO, CONTINUE
        JMP OPUT        /YES, GO TO OUTPUT PROGRAM
        CLA IAC
        DCA FLAG2
        JMS DO          /READ ANOTHER BLOCK OF DATA
        JMP CMPRES-1    /FROM DISK

/OUTPUT PROGRAM
OPUT, HLT CLA
        TAD (1777
        DCA 10
        DCA EADDR
```

```
        DCA ADDR
        JMS READ
        TAD WADD
        CLL RAL
        SZL CLA
        TAD (40
        TAD WEAD
        LSR
        4
        CIA
        DCA TALLY
        JMS FINISH
DACON,  6462
        TAD I 10
        6464                /LOAD DATA OUTPUT BUFFER
        6452                /RESET CHANNEL ADDRESS
        6111                /SKIP ON CLOCK PULSE
        JMP .-1
        6454                /D-A CONVERT
        CLA
        TAD 10
        TAD (-5777
        SZA CLA
        JMP .+3
        TAD (1777
        DCA 10
        TAD 10
        TAD (-3577
        SZA CLA
        JMP DACON
        JMS FINISH
        JMS READ
        ISZ TALLY
        JMP DACON
        HLT

*7400
        SNA CLA
        JMP .+4
        TAD OFF1
        DCA CMPRES
        SKP
        ISZ GET
        JMS FINISH

/MAIN PROGRAM LOOP FOR COMPRESSION
CMPRES, NOP
        NOP
        TAD I LIMN
        SMA CLA
        JMP .+13
        TAD 10
        SMA
        JMP .+6
        TAD (-5677
```

```
        SPA CLA
        JMP .+4
        TAD (4100
        JMP .+3
        CLA
        TAD (100
        TAD 10
        CIA
        TAD I BEGS
        SZA CLA
        JMP NOW
        TAD OFF
        DCA CMPRES+1
        IAC
        DCA FLAG1
NOW,    TAD 10
        CIA
        TAD I BEGS
        SZA CLA
        JMP .+16
        TAD I ENDS
        DCA 10
LET,    TAD I GET
        TAD COUNT
        SZA CLA
        JMP .+6
        ISZ GET
        ISZ BEGS
        ISZ ENDS
        TAD ON
        SKP
        TAD OFF1
        DCA CMPRES
        TAD 10
        SMA
        JMP .+6
        TAD (-5777
        SNA CLA
        JMP CHANGE
        DCA FLAG
        JMP .+5
        CLA
        TAD FLAG
        SNA CLA
        JMP CHANGE+2
        TAD 10
        SPA
        JMP .+12
        TAD (-2177
        SPA CLA
        JMP .+10
        TAD FLAG2
        SNA CLA
        JMP .+5
        ISZ COUNT
```

```
ON, NOP
        DCA FLAG2
        CLA

/INSTRUCTIONS FOR AMPLITUDE SMOOTHING
        TAD FLAG1
        SNA CLA
        JMP OUT
        TAD I LIMP
        TAD I LIMN
        DCA I LIMP
        TAD I LIMP
        MQL
        TAD I 10
        SPA
        JMP .+3
        JMS MUL
        JMP DRINK
        CIA
        JMS MUL              /MULTIPLICATION SUBROUTINE ON
        CIA                  /PAGE ZERO
DRINK, DCA I 11
        ISZ TALLY1
        JMP CMPRES
        TAD ON
        DCA CMPRES+1
        TAD (-101
        DCA TALLY1
        ISZ LIMN
        ISZ LIMP
        DCA FLAG1
        JMP CMPRES
OUT, TAD I 10
        DCA I 11
        JMP CMPRES
CHANGE, TAD (1777
        DCA 10
        TAD 11
        CIA
        TAD (1777            /WRITE THE COMPRESSED BLOCK
        DCA WC               /OF DATA ON DISK
        TAD (1777
        DCA CA
        TAD WEAD
        DEAL
        CLA
        JMP BACK
OFF, JMP NOW
OFF1, JMP LET

        $

/       WHEN USING THIS PROGRAM CHANGE THE CONTENTS OF
/LOCATION "VALUE" IN THE DECISION MAKING PROGRAM TO
/ "-22".
```